Professional Technician's Guide to Stereo FM Servicing

A concise presentation of practical information essential to proficient servicing of home and auto stereo FM systems.

Edited by the publishers of **Electronic Servicing** *magazine*

Professional Technician's Guide to Stereo FM Servicing

The material in this servicing guide has been adapted from a series of articles published in **Electronic Servicing** magazine.

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Chapter 1

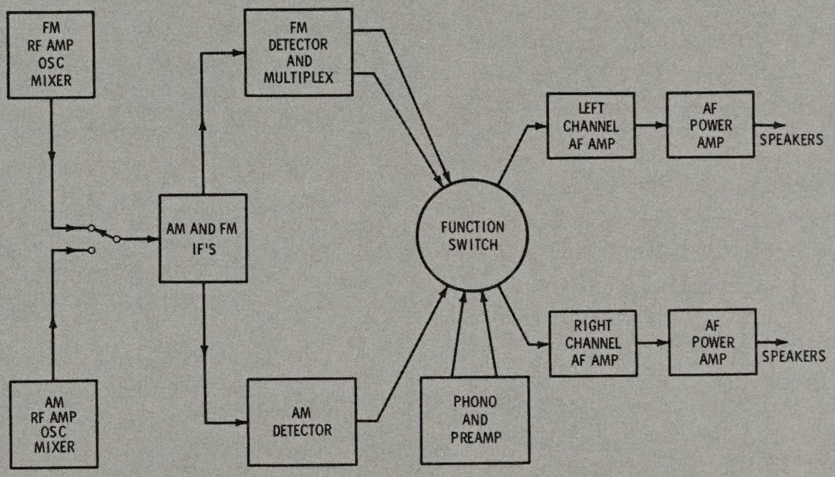
How the stereo FM signal is produced

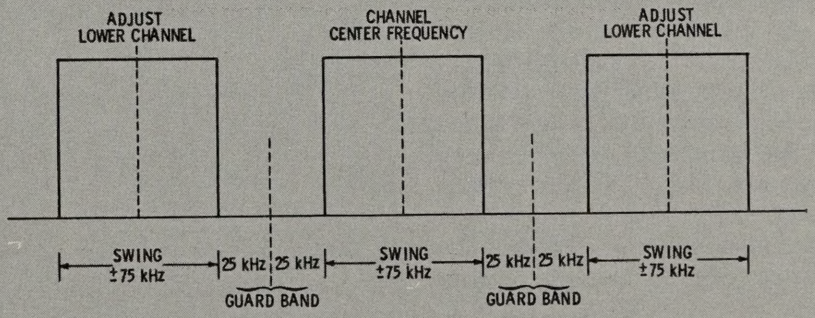
A block diagram of an FM stereo multiplex receiving system is shown in Fig. 1.

Basically, an FM-stereo signal consists of two different audio-frequency signals that occupy the same FM channel. These separate audio signals provide stereophonic sound reproduction. The individual audio signals are identified as "left" (L) and "right" (R). In conventional programming, this pair of audio signals originates from a pair of microphones at a sound studio, as depicted in Fig. 2A. The audio signal from the L microphone differs from that of the R microphone. Therefore, the stereo signal consists of two audio waveforms that vary independently in frequency and amplitude. At the receiver, the L and R signals are fed to separate speakers. These speakers are separated to simulate the placement of the transmitting microphones.

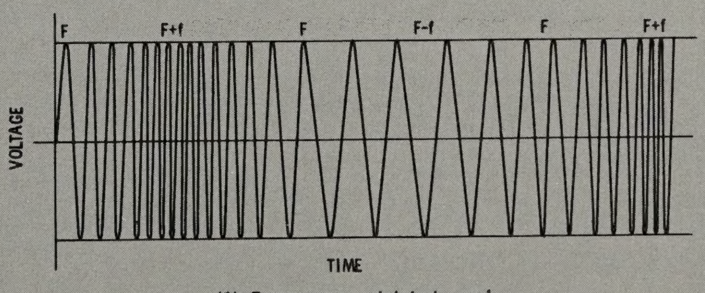
Fig. 2B shows the FM channel allocations. In monophonic transmission, the carrier (center frequency of a channel) is frequency-modulated as depicted in Fig. 2C. A swing of ±75 KHz represents FM sidebands that occupy the entire channel. In turn, the question arises concerning how one channel can be used to transmit two signals without mutual interference. If high fidelity were not required, an FM channel could be divided into two equal parts for transmission of the L and R signals on individual carriers. However, since high-fidelity transmission is a basic requirement, a method must be employed that permits each signal to occupy the entire channel bandwidth.

To transmit both the L and R signals at high fidelity in a single channel requires multiplex transmission. This is a technique which permits satisfactory separation of





(B) Frequency spectrum



(C) Frequency-modulated carrier

the L and R signals at the receiver. Note also that the system specified by the FCC is a compatible system. In other words, to a conventional FM receiver (monophonic receiver) the multiplex signal "looks like" an ordinary FM signal. But to a stereo-multiplex receiver, the multiplex signal "looks like" separate L and R signals.

A stereo-multiplex system starts with the conventional monaural audio signal, which is produced as the sum of L and R signals. That is, two microphones are employed as the equivalent of a single microphone. When the L and R signals are mixed, as shown in Fig. 3, the mono (L+R) signal is produced. This L+R signal is frequency-modulated on the RF carrier, and the result is the same as if a single microphone were used. Furthermore, to an ordinary FM receiver, only a mono signal is being transmitted—actually, as explained next, additional information to which on ordinary FM receiver is unresponsive is also being transmitted.

To explain this additional transmitted information, let us consider the effect of adding a 38-KHz carrier as depicted in Fig. 4. Both the L+R signal and the 38-KHz carrier are frequency-modulated on the RF carrier. However, only the L+R signal can be reproduced at the FM receiver. That is, the 38-KHz carrier is out of the range of audibility. Next, if amplitude modulation is impressed on the 38-KHz carrier, this modulated signal will be inaudible on an ordinary FM receiver. With reference to Fig. 5, the L+R signal is frequency-modulated on the RF carrier as before. In addition, an audio signal, A2, is amplitude-modulated on the 38-KHz carrier; this AM carrier subsequently modulates the RF carrier in combination with the L+R signal.

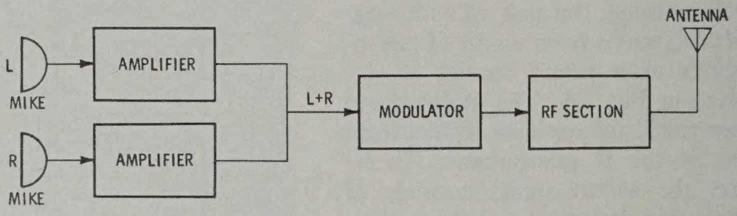
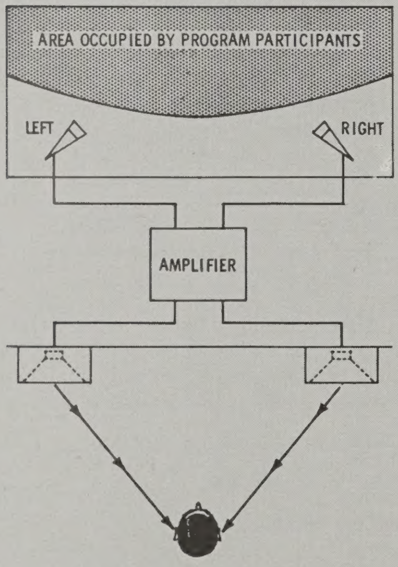
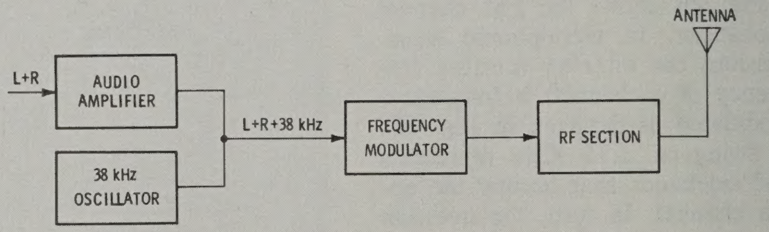


Fig. 3 L and R signals are mixed to produce monaural signal.

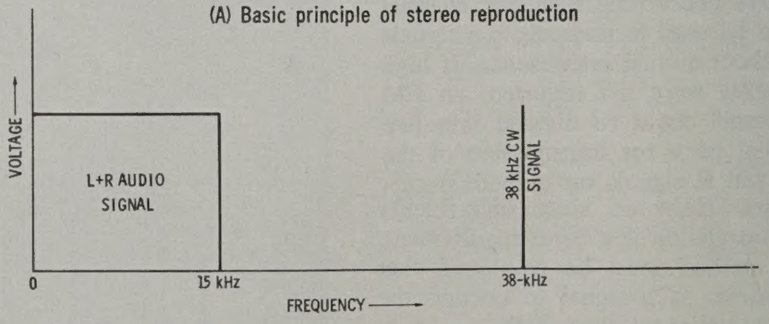


(A) Block diagram

Fig. 2 Basic concept and characteristics of FM stereo system.

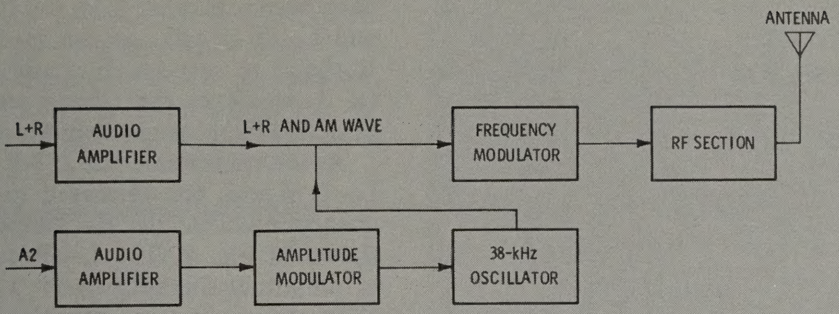


(A) Basic principle of stereo reproduction

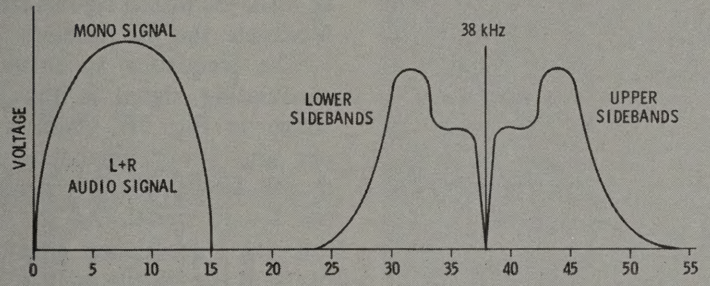


(B) FM channel specifications

Fig. 4 L+R and 38-KHz signals frequency modulated on RF carrier.

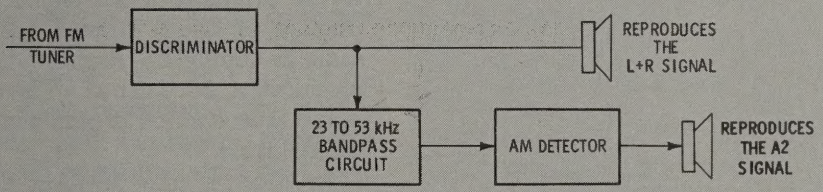


(A) Block diagram

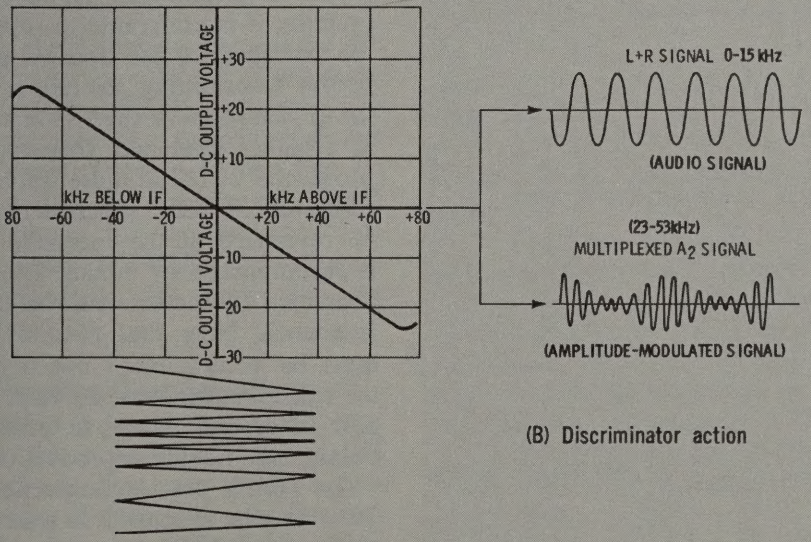


(B) Frequency spectrum

Fig. 5 Amplitude-modulated 38-KHz subcarrier combined with L and R signals.



(A) Basic system for decoding the multiplex signal



(B) Discriminator action

Fig. 6 Method for reproducing the multiplexed signal.

Fig. 5B shows the frequency spectrum of the modulating waveforms. The L+R signal has frequencies up to 15 KHz. On the other hand, the amplitude-modulated 38-KHz carrier has frequencies in the range from 23 KHz to 53 KHz. Note that after this frequency spectrum is frequency-modulated on the RF carrier and then processed through the discriminator of an ordinary FM receiver, the frequency spectrum is recovered in its original form, as depicted in Fig. 5B. Of course, only the L+R signal can be made audible. That is, the frequencies from 23 KHz to 53 KHz are said to be "encoded" in the radiated FM signal.

Now, let us see how the encoded signal can be recovered and made audible at the FM receiver. Fig. 6A shows a bandpass filter with response from 23 KHz, to 53 KHz, driven by the discriminator and followed by an AM detector and a speaker. It is evident that the bandpass filter picks out the AM signal, which is then demodulated by the AM detector. The demodulated (audio-frequency) wave envelope is then fed to a speaker which reproduces an audible signal. This is the A2 signal shown in Fig. 5. Note that the output from the discriminator in Fig. 5B is a mixture of the L+R and the A2-AM signal. Thus, the bandpass filter provides separation of the signals.

The foregoing example illustrates the basic principle of multiplex operation. However, the simple arrangement that we have analyzed must be elaborated somewhat to actually transmit R and L signals, and to obtain the R signal from one speaker and the L signal from the other. This requires the development of an L—R signal with a phase inverter at the transmitter, as shown in Fig. 7.

By inverting the polarity of the R signal and adding it to the L signal, we obtain an L—R signal. Thus we have both L+R and L—R signals available at this point. Note that if L+R is added to L—R, we can obtain 2L (the L signal with the R signal cancelled or separated). Again, if L—R is subtracted from L+R, we can obtain 2R (the R signal with the L signal cancelled or separated).

Addition or subtraction is accomplished in mixers, with associated inverters. After the R and L signals have been separated from the L+R and L—R signals, we can feed the R signal to one speaker, and feed the L signal to the other speaker, thus obtaining stereo reproduction.

At the transmitter, the L+R and L—R signals are employed to frequency-modulate the RF carrier as shown in Fig. 8. The L—R signal is amplitude-modulated on a 38-KHz carrier, and the resulting signal is mixed with the L+R signal. In turn, the mixed signals frequency-modulate the RF carrier.

The frequency spectrum of the modulating signal is the same as shown in Fig. 5B, wherein the upper and lower sidebands flanking the 38-KHz carrier are produced by the L—R signal. At the receiver, this same frequency spectrum appears at the output of the discriminator.

To obtain stereo reproduction, the L—R signal is separated by means of a 23- to 53-KHz bandpass filter, amplitude-demodulated, and further processed in a phase inverter and a pair of mixers, as shown in Fig. 9. The addition of L+R and L—R produces a 2L signal. Subtraction of L—R from L+R (the same as adding L+R and —L+R) produces a 2R signal. Thereby, stereo reproduction is obtained from the speakers.

In theory, the 38-KHz carrier (technically termed the subcarrier) could be transmitted. However, in practice, it is preferable to suppress the subcarrier at the transmitter, to permit transmitting the upper and lower sidebands of the L—R signal at a higher level and, thereby, improve the signal-to-noise ratio. In turn, the 38-KHz subcarrier must be reinserted at the receiver. This is accomplished by mixing a locally generated subcarrier with the L—R sidebands. Note that reinsertion must be accomplished not only at the exact frequency of 38 KHz, but also in correct phase, in order to obtain high-fidelity reproduction.

To permit precise reinsertion, a 19-KHz pilot subcarrier is transmitted for receiver synchronization. Fig. 10 shows how the 19-KHz pilot subcarrier is transmitted in an empty portion of the spectrum between the L+R signal and the lower sideband of the L—R signal. Interference is thereby avoided and good separation is facilitated at the re-

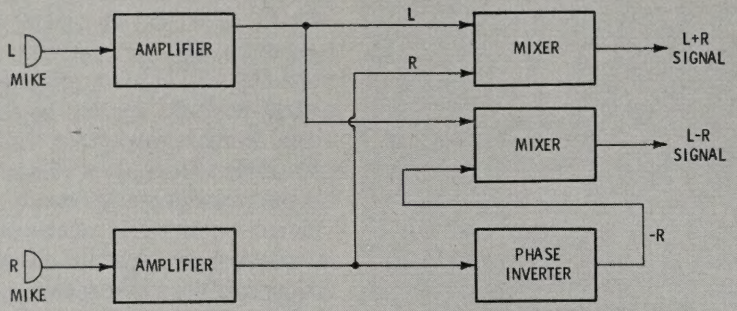


Fig. 7 Developing the L+R and L-R signals.

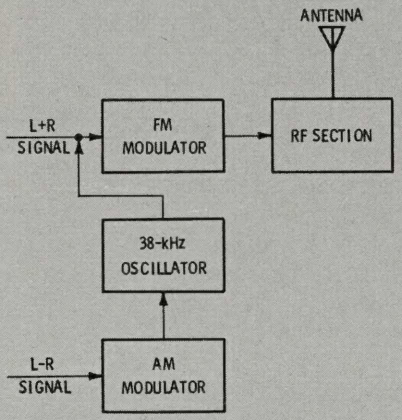


Fig. 8 Generation of the composite stereo signal at the transmitter.

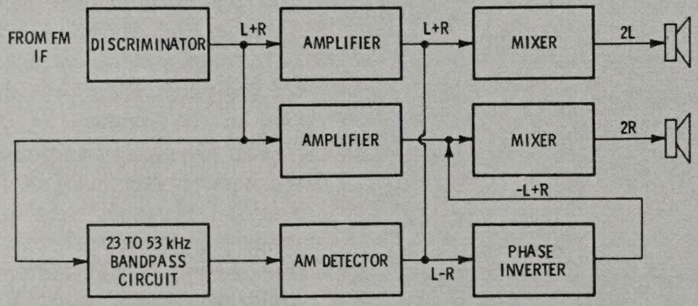


Fig. 9 Block diagram of multiplex adapter illustrates method used to obtain stereo reproduction.

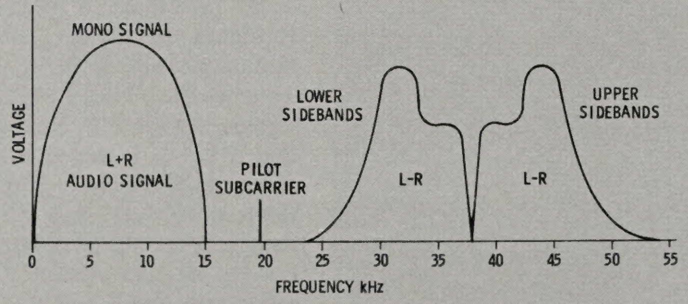


Fig. 10 Composite stereo signal showing 19-KHz pilot subcarrier in empty portion of spectrum.

ceiver. Note that the 38-KHz subcarrier is generated at the transmitter, but is trapped out before it is broadcast. A 19-KHz pilot-subcarrier oscillator is locked to the 38-KHz subcarrier signal, thereby maintaining precise phase and frequency relations.

At the receiver, a tuned circuit picks out the pilot subcarrier and feeds it to a doubler, thereby developing the 38-KHz subcarrier. This subcarrier is then reinserted by mixing it with the L—R sidebands. The end result is the same as if the 38-KHz subcarrier had been transmitted with the L—R sidebands. ▲

Chapter 2

Tuner performance evaluation and adjustments

Checking out a stereo FM tuner is a more demanding job, technically, than checking out or aligning the most complex AM radio or AM tuner. The circuitry in stereo FM tuners, especially in tuners equipped with stereo multiplex decoders, is more complex than that in AM radios. Also, reception problems are greater, and customers seemingly are more particular, since they have come to recognize FM as a high-fidelity, static-free method of broadcasting.

In the following paragraphs are briefly described some of the techniques for checking out stereo FM tuners that I have found useful over the years. Those described first require little or no test equipment. Techniques for checking the same parameters but which require more sophisticated test equipment are outlined later.

Important Test Specifications

Before you can accurately measure the performance of an FM tuner, you must have a clear understanding of the major specifications which define an FM tuner's performance. In the case of monophonic FM tuners, five major and half a dozen minor, or less important, specifications tell the performance story. The major "specs" are:

1) IHF sensitivity
2) Signal-to-noise ratio
3) Harmonic distortion
4) Drift
5) Frequency response

Less important "specs" include:

6) Capture ratio
7) Selectivity
8) Spurious responses
9) IM distortion
10) Audio hum
11) AM supression

We shall deal, primarily, with the first five of these "specs".

In addition to being able to check the performance of a tuner with respect to these specifications, a service technician should be able to align and calibrate a tuner, both RF and IF sections; adjust the AFC (automatic frequency control), if such adjustment is provided for; adjust any interstation muting circuitry; and, in the case of stereophonic FM tuners (which now are probably in the majority), measure such additional parameters as:

12) Stereo separation
13) Stereo sensitivity
14) Sub-carrier rejection
15) SCA rejection

Tests Performed Without Test Equipment

While a quantitative sensitivity analysis cannot be performed properly without test equipment, some idea of sensitivity can be obtained by comparison analysis with sets known to be in working order—or, better still, whose sensitivity figures are known—or by analysis of the customer's description of the trouble symptom(s), particularly how quickly and how much the performance changed.

FM quieting characteristics are unlike AM signal-to-noise characteristics, as can be seen in Fig. 1. As the signal strength at the antenna terminals increases, two things happen first: the audio output level increases and the noise decreases. Very soon, however, a point is reached beyond which a further increase in signal strength results in no further increase in audio level but does cause further reduction of residual noise. This point is known as "full limiting". By tuning across the FM band, and allowing for differences in program material, the experienced technician, simply by listening for large differences in audio level from station to station, can easily tell if most stations are causing "full limiting" in the IF circuits. Better sets will reach full limiting at an input signal of just a few microvolts. Less-expensive sets often require hundreds of microvolts to reach this "plateau".

Short of a defective IF amplifying stage or a totally misaligned RF or IF section, there is little that can cause a set to limit poorly, assuming that it was operating correctly in the first place. Consequently, in many cases, the trouble will be found external to the receiver. For example, the two conductors of the antenna lead-in could have been shorted together by the housewife in the course of dusting behind the set. Or, the antenna could have been disconnected from its transmission line, or the outdoor antenna could have been damaged or broken in a storm. (In a recent study, we found that a moderately sensitive tuner connected to an outdoor antenna exhibited better sensitivity and quieting than did a higher-priced tuner equipped with a 300-ohm indoor "twin-lead" dipole antenna, tossed casually on the floor behind the set.)

At least an elementary idea of sensitivity and quieting can be obtained from off-the-air measurements, using just an oscilloscope and an AC VTVM. Simply connect both of these instruments to the output of the tuner and tune to a desired station, preferably one broadcasting "voice". Set the vertical gain control of the scope so that the peaks of audio modulation just about fill the screen. In Fig. 2 we have approximated this setting using a sine wave produced by a signal generator so that we could photograph a stationery image, but the same instantaneous peak amplitude can be set up on your scope, using random modulation from program material. Next, observe the amplitude of the residual noise on the scope during moments of silence, or absence of modulation. Such residual noise is shown in Figs. 3 and 4. Notice that in Fig. 3 the noise is substantial compared with the previous peaks of modulation, whereas in Fig. 4 the noise level is noticeably lower.

I purposely did not change the vertical gain setting of my scope, to make it perfectly clear what we are trying to show, but once you get the idea, you can readily increase the vertical gain of your scope by fixed amounts (my scope has 10:1 steps, equal to about 20 dB per step). Thus, if you had to shift the vertical gain setting two switch positions to make the noise amplitude appear as great as the former modulation, you would know the noise is 40 dB **lower** than full modulation. In other words, this would represent a signal-to-noise ratio of 40 dB. Of course, in the absence of a field-strength meter or a good signal generator, you cannot establish the strength of the signal in microvolts to produce an exact signal-to-noise ratio, but you can, at least, make relative measurements from one set to another, or from one station to another, and in that way establish what represents "good" performance in your area with a given antenna installation.

By observing the modulation of a noise-free station and comparing it with that of a somewhat noisier station, you can approximate the ability of the set to go into "full limiting", even with a relatively weak signal applied, because the audio amplitude will be the same, even though the noise content might be considerably different. Again, remember, in the absence of proper test equipment, the results can be measured only in a qualitative rather than a quantitative sense.

Dial Calibration

Accurate dial calibration can be accomplished without the aid of test equipment, because, in all probability, the accuracy of even the best of signal generators will be poorer than the frequency accuracy of a broadcasting FM station, whose frequency deviation limits are set by the Federal Communications Commission (FCC). First, make certain that the dial pointer itself has not "slipped" on the dial string. If calibration is to be attempted after restringing, follow the manufacturer's instructions for setting the pointer with the rotor of the variable tuning capacitor fully closed. Usually, this will be at the lowest mark on

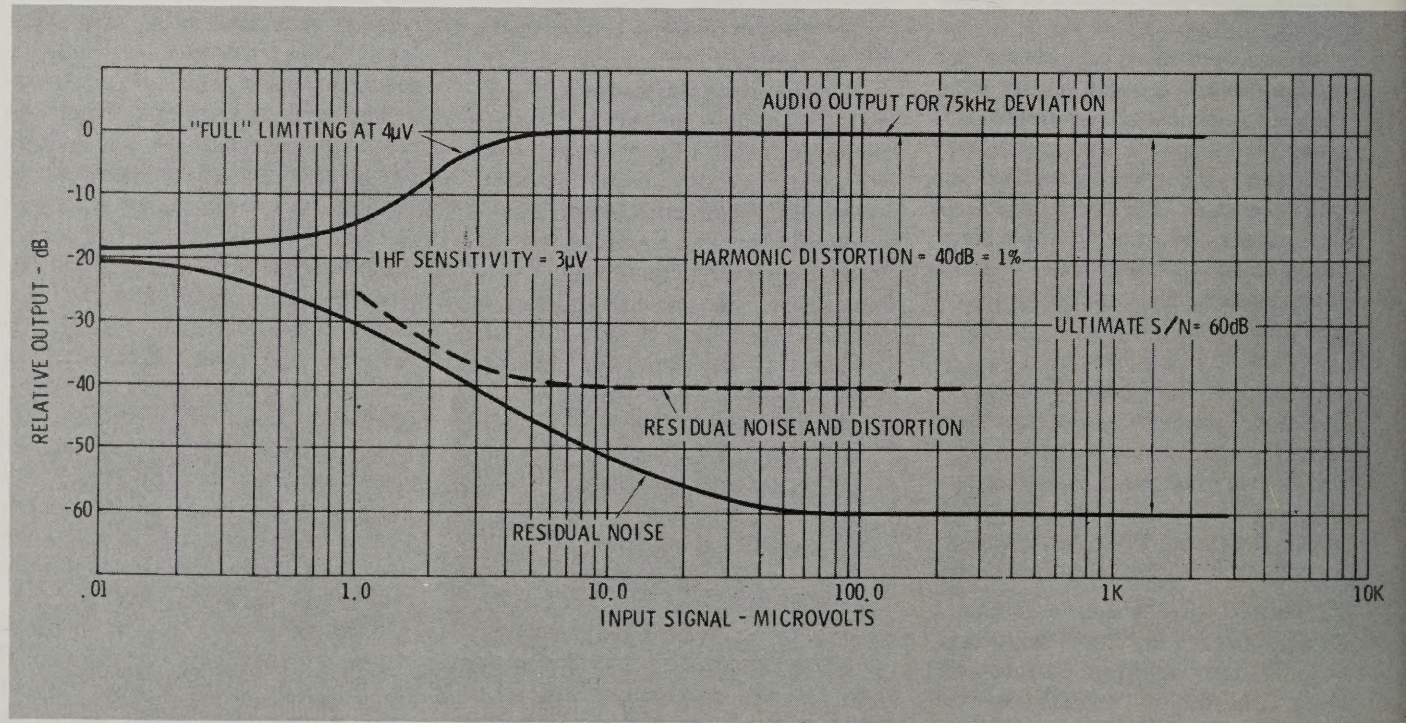


Fig. 1 Quieting and limiting characteristics of a typical FM tuner.

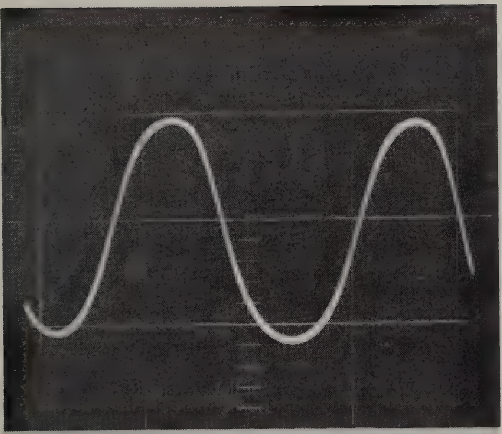


Fig. 2 Waveform amplitude produced by scope which was calibrated so that 75-KHz RF deviation (using a 400-Hz audio tone) produced full vertical deflection of the scope beam.

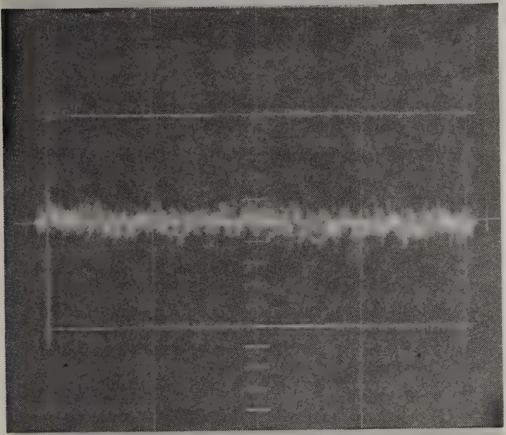


Fig. 3 Amplitude of residual noise, in the absence of modulation, can be compared to maximum audio level of Fig. 1, to approximate signal-to-noise performance of a received program.

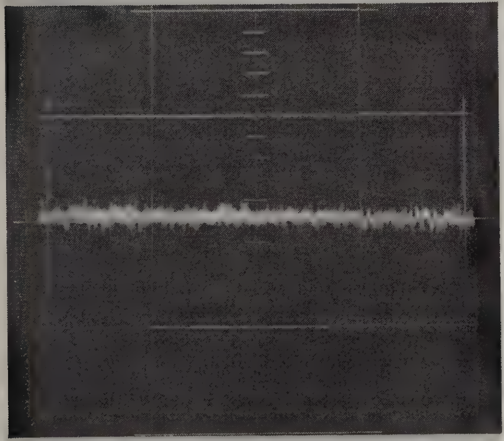


Fig. 4 Stronger RF input signal produces greater noise rejection.

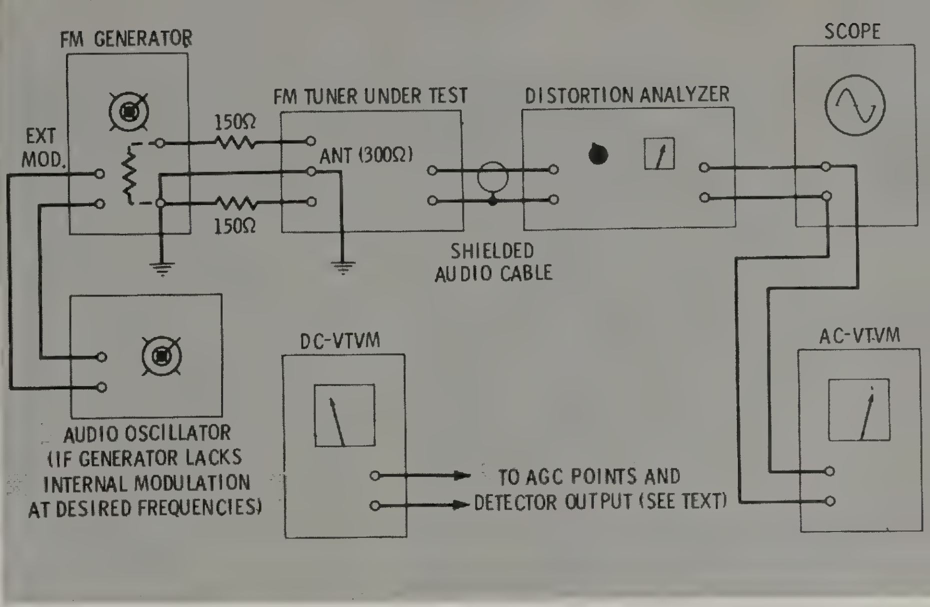

FM GENERATOR
EXT MOD.
150Ω
150Ω
FM TUNER UNDER TEST
ANT (300Ω)
DISTORTION ANALYZER
SCOPE
SHIELDED AUDIO CABLE
AUDIO OSCILLATOR (IF GENERATOR LACKS INTERNAL MODULATION AT DESIRED FREQUENCIES)
DC-VTVM
TO AGC POINTS AND DETECTOR OUTPUT (SEE TEXT)
AC-VTVM

Fig. 5 Proper test setup for making FM tuner performance measurements.

the scale, or at some extra mark or numeral indication provided for just that purpose. Cement the pointer in place on the string, using glyptal or nail polish, after first crimping the pointer onto the new string.

Next, tune in a known station at the low-frequency end of the dial (preferably at about 90 MHz) and wait for positive identification of the station by its call letters, if your FM dial is "crowded". If there is a discrepancy between the station's frequency and the dial setting, carefully tune the local oscillator coil, increasing the coil inductance to lower the frequency, if the station is tuned too low on the dial, or decreasing the coil inductance, if the station is too high on the dial. If more than a small amount of correction is necessary, you also will have to retune any antenna and RF coils located in the front end. If the set has an AGC circuit, this retuning can be accomplished best by tuning for maximum AGC voltage, using a DC VTVM as the readout device. In those sets with no AGC test point or no AGC circuit, I have found that by disconnecting the antenna, or using a minimal antenna, such as a short piece of wire, the signal can be attenuated to where it is noisy and well below limiting. With the signal thus attenuated, I then carefully "tune by ear" (contrary to all textbook admonitions) in much the same way that I have been "peaking up" AM radios for years. I tune for the loudest and the least noisy reception. I repeat these tests at a high frequency (around 106 MHz, if possible), adjusting the oscillator trimmer, and antenna and RF trimmers, as required, to reestablish correct "high end" calibration. With a little experience, you will find it almost as easy to align the front end of an FM tuner using this method as it is with a signal generator.

IF Alignment

Alignment of the IF section of an FM tuner without test equipment is a bit trickier, because the setting that produces maximum signal is not always the optimum setting. Many manufacturers prefer some form of stagger tuning, to provide better bandpass in their IF chain. In addition, in the absence of an accurate 10.7-MHz signal source, off-the-air IF alignment attempts can result in detuning of the IF circuits to some other center frequency, such as 10.8 MHz or 10.6 MHz, at which frequencies the Q's of the interstage transformers might be lower, resulting in poorer overall gain, limiting and quieting.

Often, however, noticeable distortion is caused by misalignment of only the detector stage of the IF system. Most FM receivers use a ratio type of detector, the secondary of which can be accurately returned with just the aid of a DC VTVM. To accomplish this, first find the point in the detector circuitry from which the audio (or, in the case of stereo sets, the stereo decoder) circuitry input is obtained, but ahead of any DC isolating coupling capacitors. Connect your VTVM to that point. If possible, set the scale of the VTVM to zero center and use the lowest range. Then, tune in a fairly strong station, tuning for maximum AGC reading or maximum indication on the peak-reading tuning meter, if the set is equipped with one. If, at that tuned point, the DC VTVM connected to the detector indicates anything but zero, tune the secondary slug of the ratio detector until the meter reads exactly zero. Distortion should then be minimum, and the detector tuned exactly to the "center of channel". Correcting a faulty detector circuit in this way also will serve to "balance" any AFC circuits, so that a station will be "pulled in" uniformly from either above or below the desired frequency. Failure of the AFC circuits to perform in this manner is another clue to detector misalignment.

Test Equipment— Costly but Necessary

The procedures outlined previously are quick but, at best, produce uncertain results. For more accurate measurements & alignment, other test equipment is needed. A good signal generator for the FM band costs about $700.00 new, (there are units used in laboratory applications which cost up to $2,000.00). If you're going to get serious about FM work, sooner or later you'll need one.

The correct hook-up of an FM generator and other equipment needed for making tests and measurements is shown in Fig. 5. The matching network shown in the diagram is necessary to match the usual 50-ohm output of the generator and the 300-ohm input employed in most FM tuners. As a result of its use, however, all microvolt readings of the generator should be devided in **half.** If you intend to measure only limiting, signal-to-noise ratios and frequency response, the harmonic distortion analyzer shown in Fig. 5 is not needed; but if you want to measure both IHF sensitivity and harmonic distortion, you will need this piece of equipment as well.

IHF Sensitivity

This most important of all FM tuner "specs" often is called "least usable sensitivity". It takes into account two factors: noise and distortion. For a signal to be considered listenable, the combination of these two "undesirables" must be 30 dB below the maximum audio output level. The level of input signal, usually measured in microvolts, required to achieve this result is called the **IHF sensitivity.** To determine the IHF sensitivity of a tuner:

1) Set the modulation on the generator to either 400 or 1000 Hz, and adjust for 100 percent (75-KHz) deviation. Either internal or external (using an audio generator) modulation can be used, depending upon the type of generator.
2) Tune the set and the generator to a "quiet" point around the center of the dial, and feed into the tuner's antenna terminals a signal of about 10 or 15 microvolts. (Remember, this requires a 20- or 30-microvolt signal out of the generator, because of the matching attenuating network).
3) Tune the set to produce the cleanest and largest audio signal, as observed on the scope.
4) "Null out" the signal by means of the distortion analyzer, reading the harmonic distortion **plus** residual noise as a percentage. If the reading is less than 3 percent you have exceeded the number of microvolts recommended for an IHF sensitivity measurement. If the reading is more than 3 percent, more signal strength is required to reach the level required for IHF sensitivity measurement. Vary the level of the input signal accordingly, until the "nulled" reading on the distortion analyzer equals exactly 3 percent. The number of microvolts then applied to the antenna terminals of the set equals the IHF sensitivity, and, of course, the lower the number, the better the sensitivity of the set.

You occasionally might encounter a situation in which the noise has all but disappeared from the observed waveform but a 3-percent reading on the distortion analyzer still has not been achieved. In such cases, distortion usually is the culprit, and is caused by insufficient bandwidth. The scope photo of Fig. 6 shows a typical "S"-curve, obtained at the audio output of a tuner. You can duplicate this waveform by: 1) applying the audio modulation of your generator to the horizontal input of your scope while the vertical input of the scope remains connected to the tuner output, and 2) increasing the deviation beyond the 75-KHV point previously mentioned. The linear portion of the "S"-curve should extend only about 50 KHz above and below the vertical axis. With 75 KHz of deviation applied, the top and bottom of the resulting sinewave at the output will be quite distorted, as shown in Fig. 7. If this condition exists, an increase in signal strength, by itself, will not produce the conditions

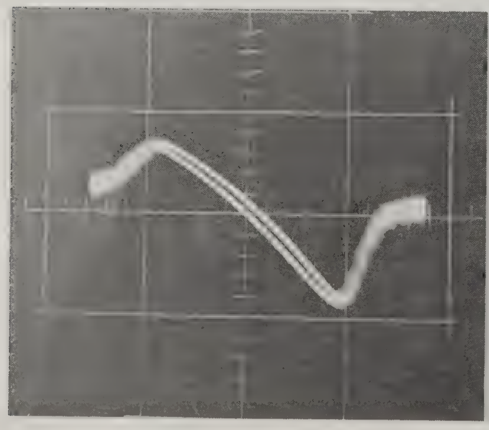


Fig. 6 Typical "S" curve observed at tuner output, using audio modulation to "sweep" FM generator carrier frequency.

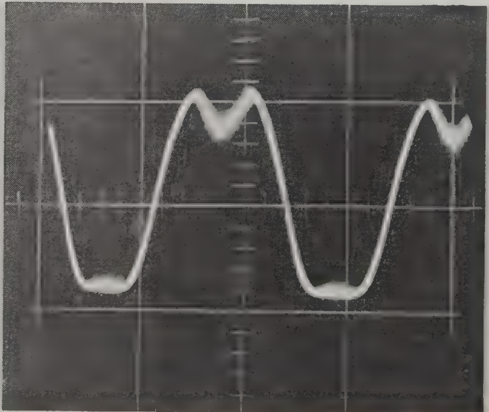


Fig. 7 Narrow bandwidth of tuner IF system produced distorted output waveform shown here.

suitable for IHF sensitivity. Usually, such distortion is caused by severe misalignment or the sudden presence of regeneration (for example, caused by an open bypass capacitor in the IF strip), which constricts, or reduces, the normal bandwidth of these circuits.

Limiting

Using the same test set-up illustrated in Fig. 5, it is easy to determine the point at which "full" limiting takes place. Simply slowly increase the input signal strength while observing the scope waveform and the meter indication, until the audio amplitude of the receiver audio output signal is within 1 dB of its highest value. The number of microvolts required to accomplish this represent the "full limiting" signal. Again, the lower the number, the better the limiting characteristics of a given set.

"Ultimate" Signal-To-Noise Ratio

The signal-to-noise ratio of a given tuner typically should increase (noise level becomes less, relative to signal level) as the level of the input signal is increased. To determine whether or not a particular tuner's performance meets this expectation, a signal-to-noise test is performed with the input signal level increased to 1000 microvolts (2000 microvolts out of the generator). The result is called the "ultimate" signal-to-noise ratio.

To determine the ultimate signal-to-noise ratio of the tuner:

1) Increase the signal input to 1000 microvolts, with full modulation applied.
2) Read the resultant audio amplitude on your AC VTVM, and observe the signal on your scope.
3) Now turn off or disconnect the audio modulation and read the (remaining) noise, hum, or or whatever extraneous signal is left. Note that the distortion analyzer is **not** required for this test. Usually, what remains will be mostly hum generated in the power supply, and this reading is expressed as the number of dB below full output. Exceptional FM tuners might have a signal-to-noise ratio of 60, 65, or even 70 dB.

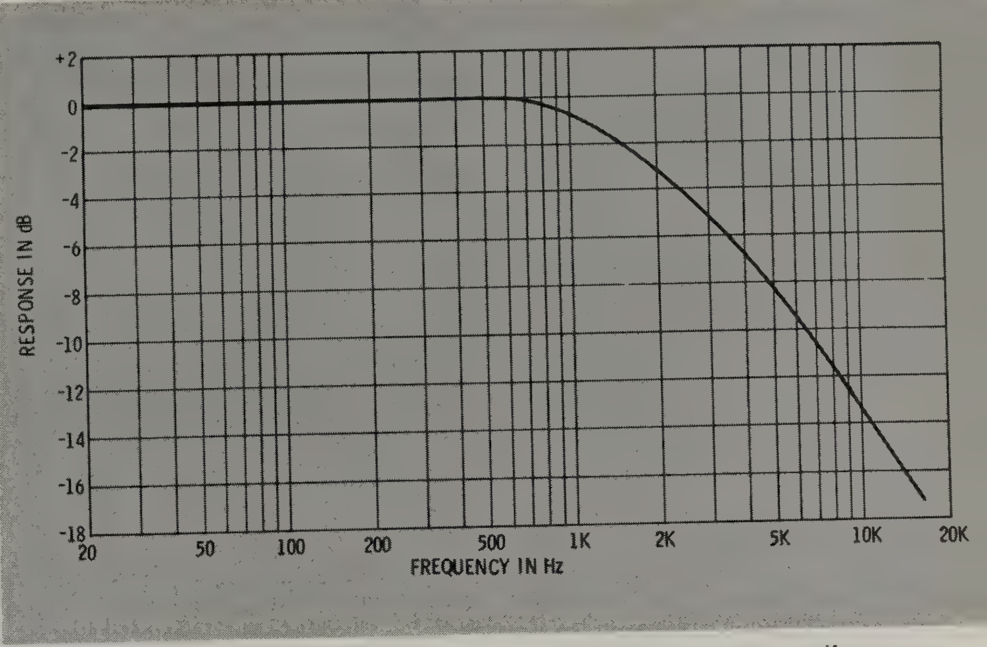


Fig. 8 Correct frequency response characteristic of an FM tuner, if no pre-emphasis is applied to modulating frequencies.

A signal-to-noise ratio below that established as normal by the manufacturer usually can be traced to excessive hum, itself usually caused by a defective component in the power supply, such as a filter capacitor.

Frequency Response

By applying to your FM generator a series of test modultion frequencies from 50 Hz to 15,000 Hz, at an RF signal input of about 1000 microvolts, and at a constant 75-KHz deviation, a plot of frequency response can be made, such as that shown in Fig. 8. Before you gasp in amazement at the "poor" high frequency response, remember that at the FM transmitter, the high frequencies were deliberately "pre-emphasized" by a prescribed amount, to produce a better signal-to-noise performance. Consequently, at the receiver end it is necessary to **de**-emphasize the highs correspondingly. Because your generator does not have this built in pre-emphasis, the output results will look like those in Fig. 8, which are entirely normal. In fact, departure from this curve in **either** direction denotes poor frequency response. If the set does not produce enough de-emphasis of high frequencies, the audio output will sound harsh or shrill.

Harmonic Distortion

Tests of ultimate harmonic distortion are accomplished exactly like the IHF sensitivity checks discussed previously, except that the signal strength at the input to the tuner is increased to 1000 microvolts—theoretically eliminating any noise contribution to the analyzer reading, leaving only the distortion to be read. Be sure to tune exactly to the center of the channel when making this measurement; even slight detuning can increase distortion considerably (a point which should be explained to every owner of an FM set). Readings as low as 1 percent or better are not uncommon with high-quality FM tuners. The lowest I have ever read was about 0.15 percent, but this is quite rare.

Alignment With Instruments

While alignment of RF and IF sections of an FM tuner does not actually fall in the category of "checkout" procedures, the use of proper test equipment for this common servicing procedure cannot be overstressed. After you've invested in a good generator, meters and scope, alignment becomes a routine and simple matter.

Because alignment procedures vary greatly from manufacturer to manufacturer, and even from model to model, only general statements about alignment can be made ac-

Fig. 9 Test setup for making stereo separation measurements, using stereo generator having RF input, or using composite audio (dotted lines).

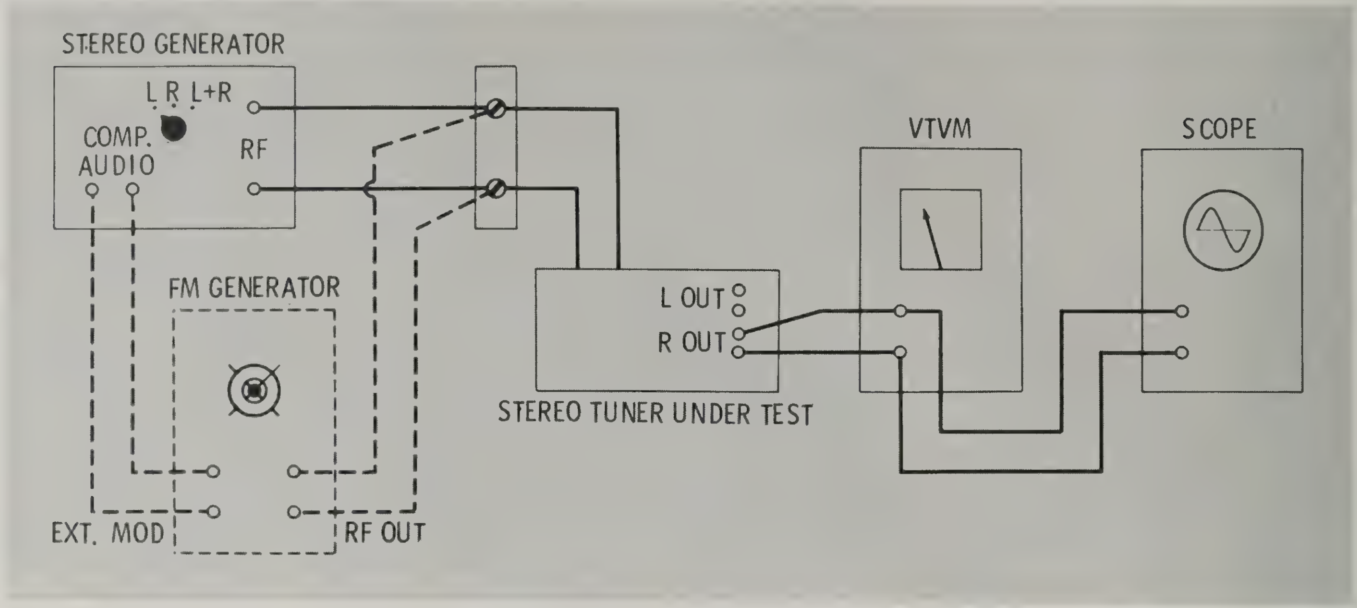
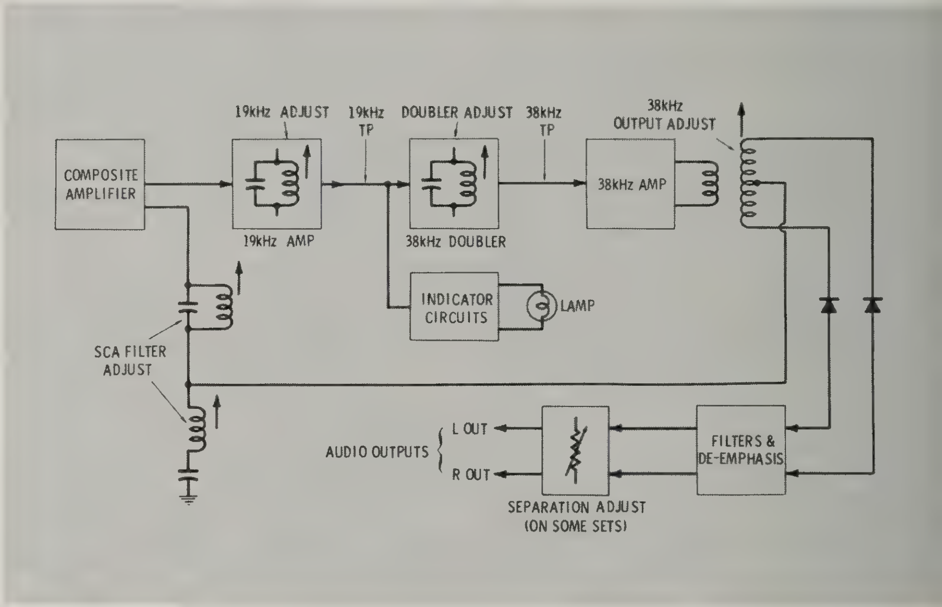


Fig. 10 Block diagram of typical stereo decoder circuit, showing adjustments and test points.



curately in this article. IF stages usually are aligned first to 10.7 MHz, using either fixed frequency techniques or sweep alignment methods, depending upon the manufacturers' recommendations. After IF and detector stages have been properly aligned, the RF section is aligned, as previously outlined, using as low an input signal level as is possible. No attempt should be made to evaluate the multiplex stereo decoder circuitry until correct monophonic alignment has been succesfully completed, because ultimate stereo performance depends, in part, upon the state of alignment of the monophonic portion of the tuner or receiver.

Separation Measurements

Proper use and understanding of an FM stereo generator or simulator will be dealt with in a forthcoming article in ELECTRONIC SERVICING. For the present, we will assume that your stereo FM generator is properly calibrated and ready for use. If it is equipped with its own RF generator as well as a composite audio output signal, it is better to use the complete generator, measuring total stereo performance of the overall tuner, rather than just decoder circuit performance. If your stereo generator has only a composite stereo audio signal output, the best procedure is to apply this signal to the external modulation input of your regular FM generator, as indicated by the dotted lines in the block diagram of Fig. 9. Because you will want to use a strong signal for basic checkout, the matching network shown in earlier diagrams is no longer essential; a slight mismatch of impedance at an input signal strength of 1000 microvolts is meaningless.

The block diagram of Fig. 10 represents a typical stereo decoder circuit. To measure separation:

1) Apply a 400-Hz, left-only signal from the stereo generator, so that it provides about 45 KHz of deviation to the RF carrier.
2) Be certain the deviation of the 19-KHz pilot carrier signal is set to the correct amount—about 6 to 7 KHz.
3) While tuning the set to produce the most and least distorted sine-wave output, read the resultant audio output on a scope and an AC VTVM connected to the **left** output of the decoder.
4) Switch the stereo generator to a right-only signal and note the drop in output amplitude at the

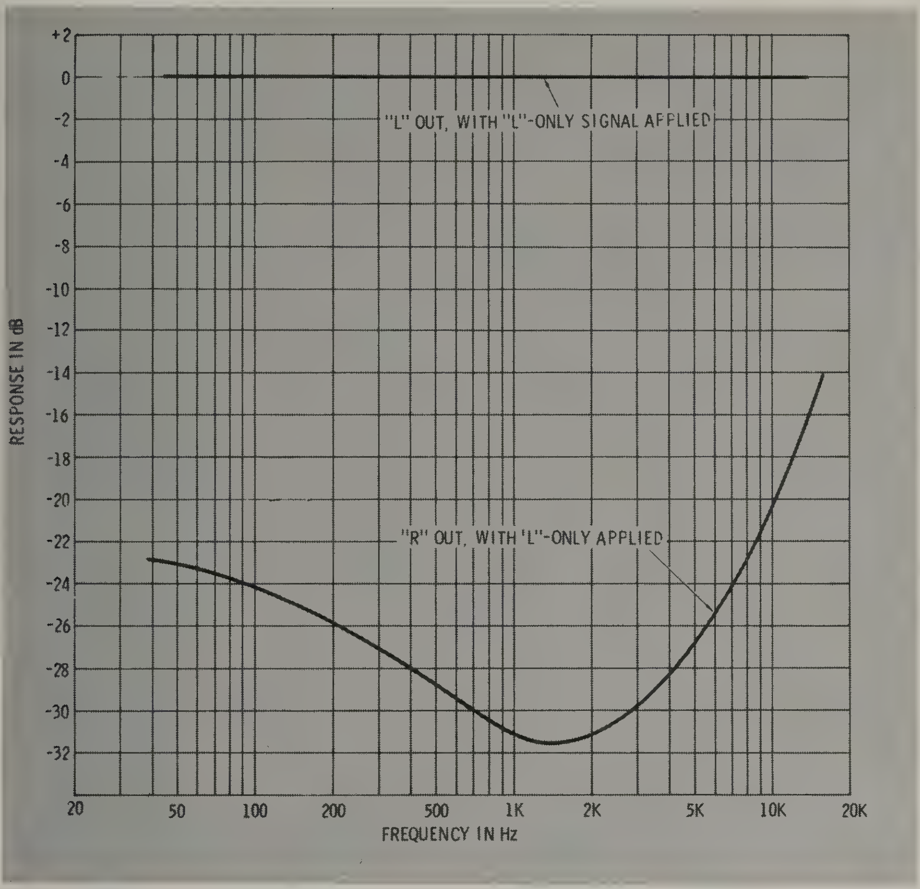


Fig. 11 Plot of stereo separation characteristics of a typical good-quality stereo decoder circuit.

left output of the decoder circuit. If this reading is greater than 30 dB, the stereo tuner has very good separation. More-typical figures for low-to-medium-quality sets generally run from 30 to about 20 dB at midband frequencies.

5) Repeat the procedure, this time applying a right-only signal and reading, first, the right output of the decoder for a reference, and then switching to a left-only signal while continuing to read the right-only output point. The two readings should be within 2 or 3 dB of each other.

Decoder Adjustments

Although the actual locations of adjustments vary from design to design, those shown in Fig. 10 represent the typical alignment points in this type of circuit. To peak these adjustments:

1) Start with the point electrically nearest the composite amplifier output, the 19-KHz adjustment. Adjust it to produce maximum 19-KHz signal, with only the pilot carrier used to modulate the RF generator. You can either monitor the amplitude of the 19 KHz at the test point shown, or, if a stereo beacon, or indicator light, is provided in the circuit, adjust for brightest indication of this light, "rocking" the slug back and forth to locate the optimum point.
2) The 38-KHz doubler tuned circuit then is adjusted **either** for maximum indication of the light (if it is located beyond that point in the circuit) or by direct visual indication on the scope and meter.
3) Adjust the 38-KHz detector transformer while observing the separation, as previously mentioned.
4) Finally, if there is a resistive separation adjustment (not present in all circuits), it should be adjusted for optimum separation. For example, often it is possible to improve the separation of the left channel, only to discover the separation has been somewhat degraded on the opposite channel. In my opinion, it is better to "compromise" the setting of the separation adjustment to produce equal separation on **both** channels, rather than superior separation on one and less separation on the other. It, therefore, is a good idea to check the opposite channel after all the adjustments in the other channel have been made, to make sure that no deterioration of the opposite-channel separation has been introduced.

A checkout of high-frequency separation is also recommended, because separation at high frequencies is often much poorer than at mid-band frequencies. A typical plot of separation vs frequency of a good-quality decoder circuit is shown in Fig. 11.

SCA Rejection Filters

Most modern stereo decoders are equipped with passive filters designed to attenuate the 67-KHz signal broadcast by stations who provide "background music" (SCA) services. If they are not attenuated by the receiver, these SCA signals can cause a "swooshing" or whistling sound in the recovered stereo program. A simple two- or three-element filter usually is sufficient to remove this interference at an early point in the decoder circuits.

To properly tune the SCA-rejection filter:

1) Apply a 67-KHz signal, with about 10-percent modulation, to your generator.
2) While monitoring the 67-KHz output at the SCA test point shown in the block diagram of Fig. 10, adjust the SCA filter circuits for a "null". That's all there is to it.

Conclusion

FM receivers and tuners are more complex pieces of equipment than their AM predecessors. The addition of multiplex stereo decoder circuitry further increases this complexity. An understanding of related specifications and methods of measuring them, as outlined above, will enable you to perform stereo FM servicing as confidently and competently as you do TV and AM servicing. ▲

Chapter 3

Localizing trouble to the tuner or the amplifier

Trouble in a high-fidelity component system can occur in either the tuner or amplifier and produce the same audible symptoms. Others are distinctly tuner trouble, and still others are always amplifier troubles. A bit of logical reasoning plus the tips suggested here should reduce your troubleshooting time to the minimum possible.

When most home hi-fi systems consisted of a separate tuner, a preamplifier and a power amplifier, it was a fairly simple matter to troubleshoot these systems by the process of elimination. For example, if FM reception failed, but the phonograph section worked, you knew positively that the separate tuner was the cause of the trouble, etc.

Today, the trend in hi-fi stereo equipment is almost exclusively in favor of complete "receivers" (tuner/preamplifier/amplifiers all built on one chassis) or even more consolidated "compact" systems, in which even the record changer is integrated within the one-piece electronic chassis. Functionally, of course, today's chassis is still divided into sections somewhat similar to those of a decade or two ago, but isolation of the offending section becomes a little bit more sophisticated, since we now are dealing with sections common to the whole system (power supply, selector switches, etc.) as well as with isolated circuits and components.

The block diagram in Fig. 1 shows the typical design of all-in-one stereo receivers; with its aid, and using much the same logic you would use in troubleshooting individual components, we shall show how it is still possible to track down troubles in this type of system using the process-of-elimination approach.

Recognizing that **some** enthusiasts of hi-fi still prefer the "separate-component" hi-fi system, we also shall examine problems caused by mismatching separate components.

Harmonic Distortion— Amp or Tuner?

An FM generator, a scope and an audio oscillator are the only test instruments needed to determine which circuit is introducing distortion in a stereo FM system. The FM generator is connected to the antenna terminals, and modulation is set to 75 KHz (100 percent) at an audio frequency of either 400 or 1000 Hz. A scope is connected to the output of the stereo decoder circuitry (either the left or right output, since a mono signal is being used), before the signals are fed on to the preamplifier/amplifier stages.

One form of distortion common to FM circuits is illustrated in the scope photo of Fig. 2. The cause of this distortion is insufficient bandwidth or detector linearity. A waveshape such as this could **never** be caused by amplifier circuits, which would tend to cause sharp **clipping,** as illustrated in Fig. 3, rather than a "dip" or valley at the tops and bottoms of the sine-wave, as in Fig. 2. Normally, bandwidth of a tuner becomes restricted in this manner in the absence of sufficient signal strength or, what amounts to the same thing, improper RF or IF alignment. If increased signal strength tends to clean up the sinewave of Fig. 2, you can be sure that the cause of the distortion is in the tuner circuits. You then should check alignment **and** antenna connections and installation. Today's tuners are so "quiet" that a weak signal can cause distorted output without the random noise accompaniment usually associated with weak-signal conditions.

On the other hand, waveshapes such as that of Fig. 3 will seldom be caused by tuner malfunction and are more likely to be the fault of amplifier circuits which are overloaded. A partly shorted speaker connection (frayed lead shorting across the speaker terminals) can reduce power output to the point where the sound is barely audible, and at the same time limit maximum clipping to produce the output shown in Fig. 3. The output of an audio generator fed to the actual amplifier input (in this case of a receiver, use the AUX input) will quickly determine if this condition exists. Apply a signal no greater than that called for in the instruction manual; most auxiliary inputs require about .5 volt for full amplifier output. Full amplifier output usually will be that power output which results in less than one percent harmonic distortion—an **amount of distortion barely visible** on a scope presentation (and barely audible, for that matter).

If no distortion is evident when conducting the preceding test, and if no distortion was detected at the output of the tuner section itself, there must exist a mismatch between the tuner and the amplifier. This often happens when a separate tuner and amplifier are used. For example, if the tuner is improperly connected to a low-level input on the amplifier (such as tape head or magnetic phono), the first preamplifier stages of the amplifier might be overloaded by the excessive input signal, though the input signal itself is not distorted and the amplifier is functioning perfectly. To a lesser degree, connecting a tuner whose output is one volt or more to an amplifier input designed to process a signal no greater than 0.5 volt also can cause noticeable distortion, but to a lesser degree.

In the case of "integrated" receivers (tuner/amplifiers), such mismatching is not likely, since the manufacturer has seen to it that output and input levels of the circuits are compatible.

High Hum Level

Most hum problems in high-fidel-

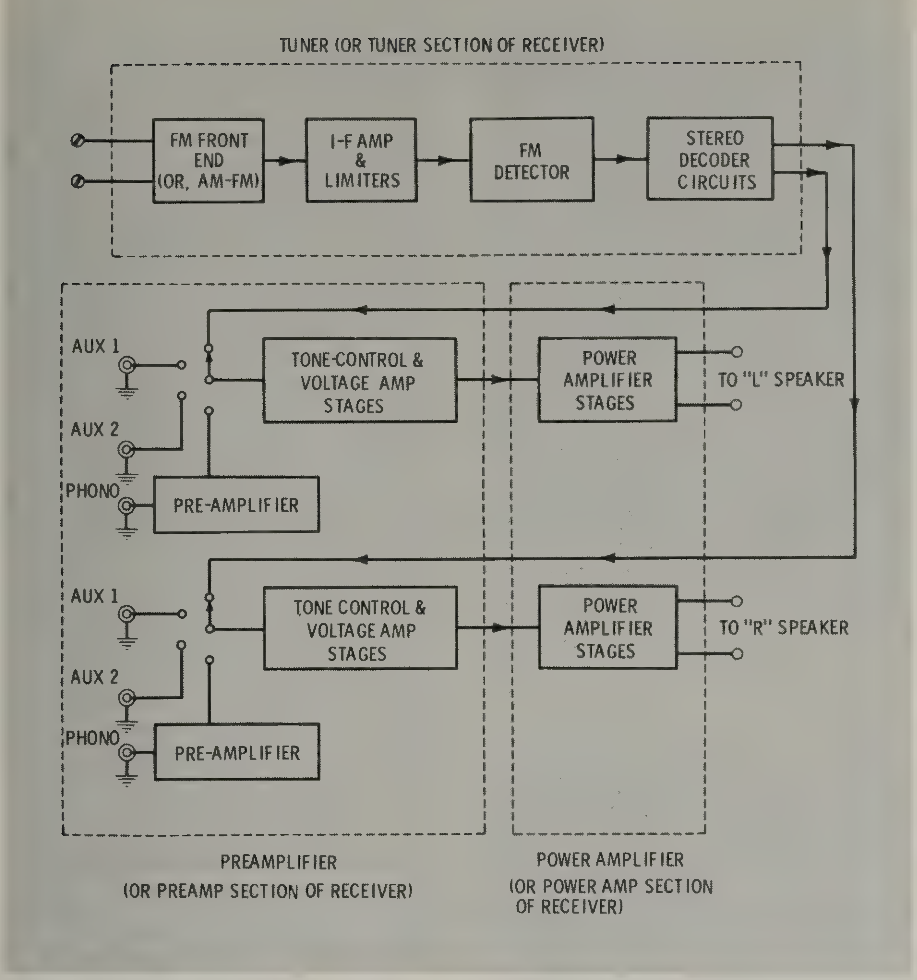


Fig. 1 An "integrated" stereo receiver, such as the one diagrammed here, can be analyzed in terms of its separate "component" functions.

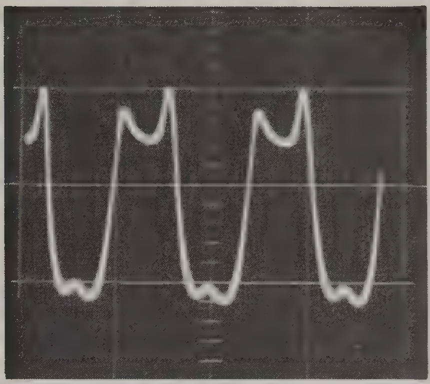


Fig. 2 Distorted sine-wave shown is the result of compressed bandwidth in the IF section of the tuner.

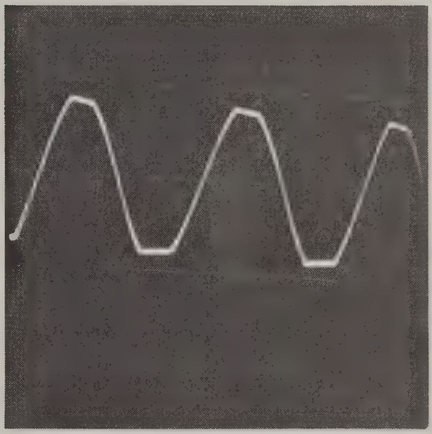


Fig. 3 "Flat-topped" clipping of sine-wave, shown here, is the result of amplifier overloading—not the fault of tuner distortion.

ity component systems occur when magnetic phono cartridges are used. The gain of amplifiers used with magnetic cartridges needs to be about 40 dB greater than the gain in use when FM circuits or other high-level signal sources are applied. Still, there are instances in which hum during FM reception can be a problem.

In separate tuners and amplifiers, hum can be induced in the AFC circuits of the FM front end by electromagnetic coupling of high hum fields. For example, I have encountered tuners mounted directly above the power transformer of an amplifier, and the hum field from the transformer was sufficient to induce hum modulation in the local FM oscillator via the AFC circuits.

If hum is present in all positions of the amplifier's selector switch, the fault is not in the tuner, but in the amplifier. For this check, don't use the phono position as a reference, but choose a high-level position of the function switch, such as "TAPE IN" or "AUX", whose impedance and gain levels are about equal to that of the tuner input.

Shielded cables between tuner and amplifier should, of course, be examined for open ground connections, or even high-resistance grounds caused by poor soldering at the shell of the pin plugs used with these cables. If removal of the interconnecting cable at the amplifier end stops the hum, suspect the cable or the tuner itself as the source of the trouble. Using a very long interconnecting cable sometimes will induce hum, even if the cable is shielded, because the shielding used in inexpensive audio cable is usually only about 60 to 70 percent effective. If the output of the separate tuner is not fed from a cathode-follower circuit (or an emitter follower, in the case of solid-state equipment), the source impedance will be high, and external hum fields present might cause sufficient voltage drop across the interconnecting cable to produce audible hum.

In the case of all-in-one receivers, the only causes of tuner-section hum I have ever encountered are an open capacitor associated with the power supply that feeds operating voltages to the tuner, and an open RF bypass capacitor in the RF section of the FM tuner.

High Noise Level

Fig. 4 illustrates the relationship between signal and noise in a typical FM tuner. Note that as the input signal strength is increased, the noise level becomes lower and lower, finally reaching a low figure which remains constant. At the same time, the audio output of the tuner section increases in amplitude for a short time with increasing signal, reaching a constant value much sooner than the residual noise reaches its final, low value.

A tuner which exhibits good limiting characteristics will reach this constant level of audio output with just a few microvolts of RF signal applied, while an inferior product might require 100 to 500 microvolts of input signal before the audio

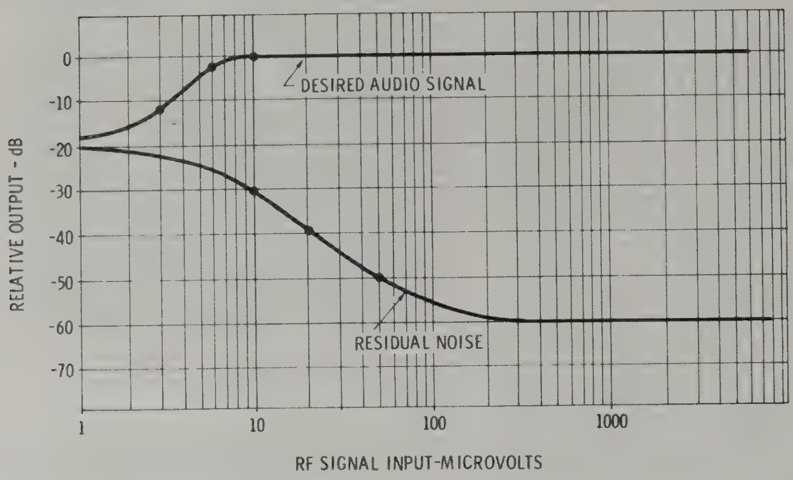


Fig. 4 FM background noise and audio output levels are a function of input (RF) signal strength, and follow the general relationships shown in the curves.

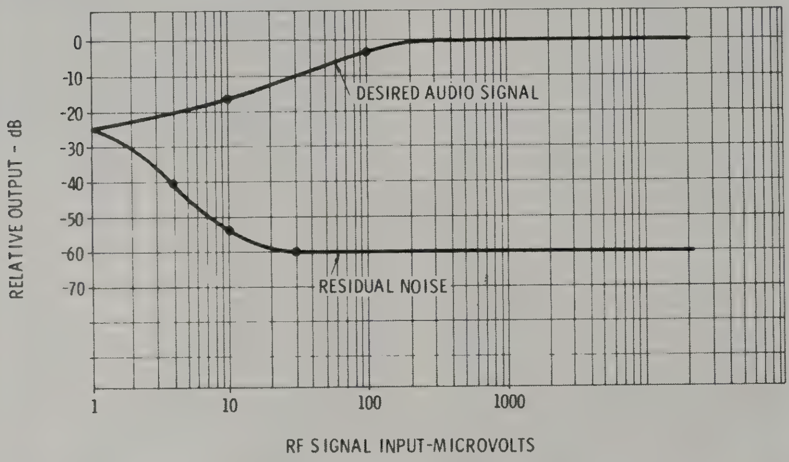


Fig. 5 In some tuner circuits, adequate "quieting" is reached at signal levels which are too low to provide "full limiting", or "full audio recovery".

"plateau" is reached. This explains how a weak signal can produce background noise, even though the audio level appears normal. For example, note that while a signal of 10 microvolts will produce full output of audio in the example of Fig. 4, the noise at this signal input level will be only some 30 dB below full output, and, therefore, quite audible and annoying.

If noise is traced to the tuner itself (by checking for other noise forms in the phono or other amplifier input positions and noting its absence), you probably will be dealing with a signal strength problem (although resistors and transistors in these circuits can produce noise). Again, this will require investigation of antenna connections, the condition of the antenna (is it rusted, are transmission line connections good, etc.?) as well as the alignment of the tuner. If the set under investigation is equipped with any kind of signal-strength meter, it can be used to check the strength of the received signal. Using an FM generator having an output calibrated in microvolts, you can easily calibrate the tuner's own signal strength meter to read directly in microvolts. For example, if you establish that 20 microvolts of input signal from your generator will cause the tuning meter to deflect two divisions, and a radio station signal causes a meter deflection of only one division, you know that the received signal is considerably less than 20 microvolts—too weak to produce quiet operation.

Background FM noise is "white noise", or noise that contains all audible frequencies. Noise produced by noisy transistors, resistors and tubes usually has a more characteristic "shot" noise effect; with practice, you might be able to distinguish between the two types just by listening. In the event that you do decide that a noisy transistor is at fault, it usually will be one located in the audio portion of the equipment, since transistors designed for use at high frequencies (such as the 10.7-MHz IF and 100-MHz regior RF associated with FM) are generally much lower-noise devices to begin with, and are subjected to greater quality control with regard to this parameter.

It's Not Loud Enough

Fig. 5 shows that under certain conditions a signal can be received with adequate quieting (absence of background noise) but so far below limiting that it is necessary to turn up the volume control more than for other received signals. This will occur in weak signal areas when you are dealing with a set that has very poor limiting. It has been my experience that many foreign-made sets suffer from this defect, because the designers place greater emphasis on noise quieting than on good listening.

Obviously, you cannot redesign these sets, but you can determine, as before, that the signal is weak and then do something to improve the audio recovery, such as installing a better antenna or, if no antenna is being used at all (other than a short piece of wire or a so-called line-cord-coupled antenna), strongly recommend to your customer that a suitable FM antenna be installed.

In the case of separate tuners connected to separate amplifiers, make certain that the low FM level is not being caused by a misadjustment of the level control usually supplied on the tuner. This control might or might not be a front-panel control. Often, it is a rear "screwdriver" control intended to be set properly during installation.

Some elaborate preamplifiers, and even integrated amplifiers, also are equipped with input-level controls used to set up equal levels from all signal sources. If the amplifier in question is so equipped, the setting of this control might have been moved accidentally and should be checked. Of course, if levels of associated signal sources, such as phono and tape, are suddenly lower than normal, the fault might be traced to the amplifier circuits, using normal audio signal-tracing techniques. Stereo amplifiers are actually an aid in this sort of tracing, since, in most cases, only one channel of the pair will exhibit loss of gain (unless the common power supply is at fault). By comparing the gain at similar points in the circuit of one channel with that of the other channel, the point of gain-loss is quickly determined.

Intermittent FM

The bane of the serviceman's existence, the elusive intermittent, occurs in stereo tuners just as it does in every other piece of consumer electronic equipment. Don't blame the FM tuner, however, until you have checked the amplifier for intermittents by playing through it a signal source other than FM. A good way to isolate the intermittent is to feed a modulated FM RF signal in at the antenna terminals and observe the scope waveform at the detector output of the FM tuner. A diagram of this setup is shown in Fig. 6. If the intermittent does **not** show up during this observation (while tapping the chassis, applying heat or coolant to the suspected part, etc.), it is reasonable to assume that the trouble is beyond this point, either in the stereo decoder circuitry (which is usually "in-circuit", even in the "mono" mode of operation) or in an early audio amplifying stage in a complete receiver. If it **does** show up, the next step is to feed a modulated 10.7-MHz signal to the 1st IF stage, using a suitable coupling capacitor. If the intermittent disappears under these conditions, the trouble is probably in the RF or oscillator stages of the tuner. A suspicious element is always the variable capacitor, whose rotor might be touching the stator at one or more points during rotation. If the intermittent still is present, the IF section should be suspected; the elements most likely to be at fault are interstage transformers, which can be internally shorted or so close to being shorted that a slight movement of the chassis completes the short.

It Must Be The Tuner

Some common defects in a high-fidelity stereo system immediately can be pinned down as being the fault of the tuner section (in a tuner/amplifier receiver) or the tuner (in a separate-component system). These include loss of sensitivity, drift, an undue amount of background noise when listening to stereo FM, and permanent or intermittent loss of stereo separation when listening to stereo FM.

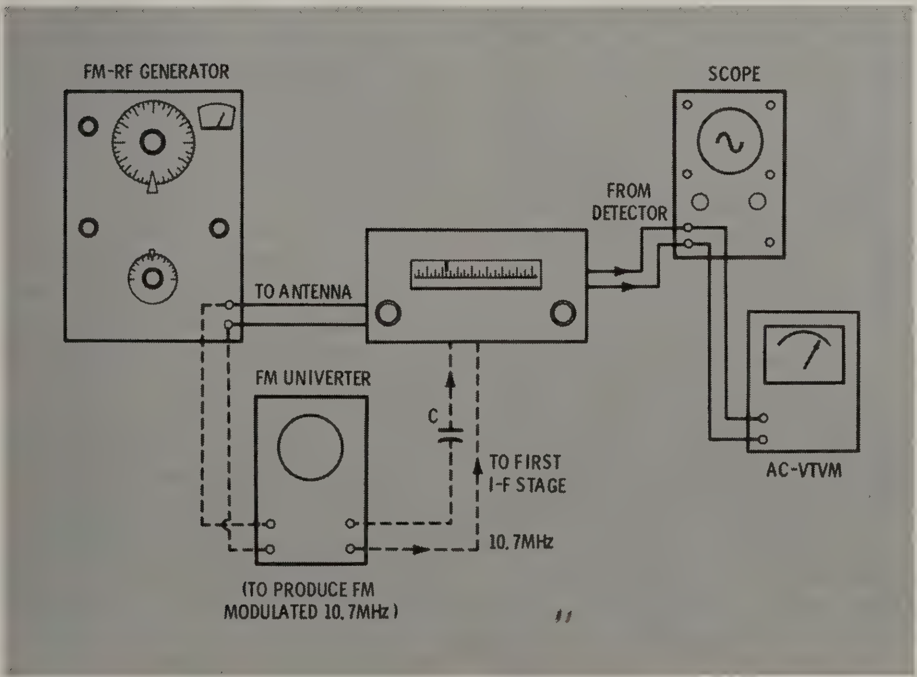


Fig. 6 Test setup for analyzing performance of tuner section in a stereo high-fidelity system.

Once loss of sensitivity has been established as **not** being caused by a broken antenna connection or a damaged antenna, look in the RF section of the tuner. A weak RF stage (tube or even transistor) will not cause complete absence of reception but will, in most instances, result in a weak, noisy FM signal. Often, an IF stage that is defective will cause the same effect.

Isolation of the exact stage at fault can be accomplished by conducting stage-gain measurements, but these measurements are a bit too complex for some technicians, and require very careful metering and scope observations.

An indirect way to measure early stage gain in a tuner or receiver is by means of the AGC circuits of the tuner or receiver. The setup is shown in block-diagram form in Fig. 7. If, for example, it takes 100 microvolts of RF signal input to produce 2 volts of AGC voltage, and it takes 1000 microvolts of 10.7-MHz IF signal applied to the 1st IF stage to produce the same 2 volts of AGC, then the gain of the "front end" must be 10, or 20 dB of voltage gain.

Similar measurements can be made from stage to stage, using a fixed amount of developed AGC voltage as a reference. If, when moving the signal generator from one stage to the succeeding one, no change in signal input is required to maintain the same AGC voltage, that stage is not producing any gain. Of course, such measurements cannot be used for stages which follow the point from which AGC voltage is derived, such as the last IF or limiter stage, but if you've gotten that far and no stage has been suspected, the last stages can be investigated by process of elimination.

Invariably, a relatively large amount of noise on stereo FM (compared to mono FM) is **not the fault** of the tuner, although it usually is hard to convince your customers of

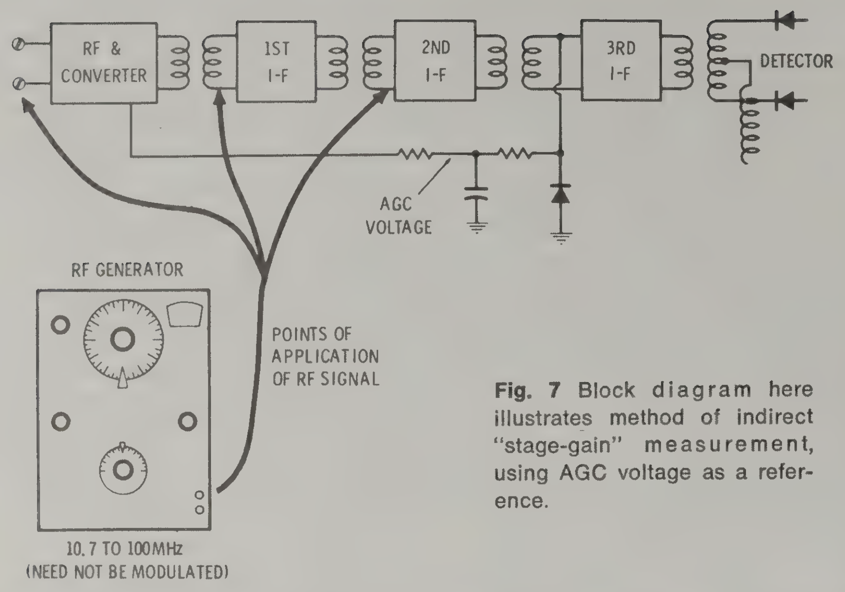


Fig. 7 Block diagram here illustrates method of indirect "stage-gain" measurement, using AGC voltage as a reference.

this fact. The truth is that stereo FM reception is about 20 dB noisier than monophonic FM. A design that produces 60 or 70 dB of signal-to-noise on a strong mono FM signal usually will produce about 40 to 50 dB of signal-to-noise in stereo FM, which is still quite acceptable. But, if a particular design produces 40 or so dB of quieting on a mono signal (which is acceptable), when it is switched to stereo FM, it will yield a 20 dB signal-to-noise ratio, which is definitely **not** acceptable. Since there's nothing you can do to the set design to increase its basic sensitivity, and since you can't ask the customer to move closer to the broadcast station, your only alternative is to sell the customer on the idea of a good outdoor FM antenna suitable for his location and distance from the stations he wants to hear. It's a lot easier (and quite common) to get an increase of signal strength of ten-to-one by using a good outdoor antenna than it is to increase tuner sensitivity by even two-to-one!

Usually, loss of separation in stereo FM is also the result of a weak signal, though other causes, such as faulty stereo decoder or switching circuits, can produce the trouble. Most modern stereo tuners have an automatic switching circuit which changes the reception mode from mono to stereo when a stereo signal is received. Most of these circuits have a "threshold" adjustment, set at the factory to switch the receiver over to the stereo mode at a given signal strength. This is done because most listeners will not want to listen to stereo FM if it is accompanied by an excessive amount of background noise. Your customer might prefer separation—even if it is noisy, so check the stereo threshold adjustment, if the tuner is equipped with one. Of course, if separation is lost for **all** stereo FM listening conditions, the trouble is more serious and the actual multiplex circuits need to be checked (**after** you make certain the customer has not accidentally set a remote mode switch to the "mono" position!).

Gradual drift is almost always caused by heat—heat which reaches frequency-sensitive capacitors, coils and active devices in the local oscillator section of the tuner. Although today's solid-state designs have decreased the probability of drift by reducing heat build-up in the chassis, placing a solid-state tuner directly over a heat-producing amplifier (even a solid-state model) can upset the carefully compensated-for circuits, and can cause drift in even the most modern set.

Tube-type sets should be given plenty of ventilation and air space around them, so that the greater amount of heat produced in these chassis will not cause excessive drift.

Unusual or severe cases of drift usually can be traced to a defective compensating capacitor associated with the local oscillator circuit of the FM tuner. The active oscillator element (transistor, FET, etc.) also might be at fault, and its voltages should be checked both when cool and an hour or so later, after the set has reached operating temperature. Another remote, but possible, cause of drift in an FM set is extreme change of power line voltage (caused when all the air conditioners are turned on in an inadequately wired house, for example).

It Must Be The Amplifier

A dead stereo channel usually means a dead amplifier channel, but not always. The easiest check is to feed a signal into the AUX input of the amplifier, using both left and right inputs. If the previously dead channel is still inoperative, the trouble is in the amplifier.

As mentioned previously, transistor noise can be mistaken for tuner noise by all but the most experienced listener. Actually, there is a difference. Transistor noise usually (but not always) is laden with a "shot" effect, and usually is not uniform in intensity. Tuner background noise is uniform, wide-spectrum noise, and is not as offensive to the listener. Still, if you're not sure, switch to AUX once more. If the noise is still there and similar in character to what you heard when the function switch of the set was in the tuner position, the trouble probably is in the amplifier.

Overheating of amplifier output transistors, blowing of fuses and popping of thermal circuit breakers are all symptoms of an overloaded amplifier. If these failures occur when listening to FM at normal levels, the trouble is in the amplifier and not in the tuner. A check for partial speaker lead shorts or unaudible oscillations, which might be driving the amplifier far into overload without anything being heard, should quickly disclose the source of the trouble. In some class AB or class B transistorized amplifier circuits, one of the output pair can be defective and the amplifier still produce sound at low or medium levels. However, in such cases, current drain might become excessive because of unbalanced biasing, producing the overload symptoms mentioned previously. ▲

Chapter 4

Tracking down separation problems

How Stereo Separation is Achieved

Some fifteen years after the popularization of stereophonic sound reproduction in home music systems, there is still much that is misunderstood about two-channel, or stereo, sound by listeners and technicians alike. The nature of the "stereo illusion" is still being intensively investigated by top scientists in the field.

Before you can logically attempt to analyze "stereo faults" in a high-fidelity stereo system, it is important to understand the few elements that go to make up what we call the "stereo effect".

The easiest phenomenon to understand about stereo is "separation". If two microphones were used to record the original program, each microphone "heard" a different program because it heard it from a different vantage point. The left microphone, in the case of a typical symphony orchestra arrangement, will pick up the string section with greater intensity than, say, the brass section, while the converse will be true of the right-channel microphone. That is not to say, however, that the right microphone picks up **no** sound from the string section or that the left microphone "hears" **no** trumpets. What we can safely say though, is that the right microphone will pick up the string section with considerably diminished amplitude, and at a somewhat later point **in time.**

When listening to the recorded results, your ears and that part of your brain which translates into "intelligence" what the ear picks up will work with these two differences between channels—intensity **and** time of arrival (which may be looked upon as "phase" differences, since at all but the highest tonal frequencies, the difference in time of arrival will be but a fraction of one full wavelength). As implied, the job of any stereophonic amplifier is to faithfully reproduce both the amplitude and phase differences between the recorded material of both channels.

Manufacturers and technicians, as well as many users of equipment, have been conditioned to adopt the "purist" attitude that the **more** electrical separation you have between channels, the better the system. Thus, claims of 25, 30 and even 40 dB of separation are made (and met) by various pieces of equipment on the market.

The truth is, however, that in all but the most exaggerated stereo recordings (in which, for example, the two channels were actually recorded in two separate studios and even **on two** separate occasions) it has been proven that virtually all listeners cannot detect the difference between 10 or 15 dB of separation or cross-talk. By that we mean, it is better to have a balanced cross-talk situation (left to right and right to left) of 12 dB than to have, say 20 dB of cross-talk from left to right and only 8 dB of isolation from right to left. This condition, which is sometimes present in mis-adjusted stereo/FM circuits, causes listeners to try to re-balance the "center" sound (with no success, by the way) by means of the balance control on the amplifier.

Speaker Placement for Stereo

Ignoring any electrical problems, for the moment, it has been our experience that most complaints from customers involving "poor stereo effect" stem from improper speaker and/or listener placement.

Equipment manufacturers are often at fault here, because, in an effort to generalize, they make such statements in their instruction booklets as :"Place the speakers 8 to 10 feet apart for optimum stereo effect." Now, 8 to 10 feet of separation might be great for certain size rooms and with the listener in an optimum position, but in other situations the sound might fuse together in what sounds little better than a two-speaker monophonic system, while in other circumstances the apparent separation might be so extreme as to sound like two separate programs with a big "hole-in-the-middle".

Over the years, we have formulated a set of basic layout dimensions which work satisfactorily. These layout dimensions are tabulated in Table I, and are based upon the listener being positioned at distances shown. Of course, the listener cannot always position himself at the optimum listening point, and adjustments have to be made. For this purpose, the graph of Fig. 1 shows the necessary correction factors to apply to the table if the listener has to sit further away or closer to the plane of the loudspeakers. Some flexibility is, of course, permitted, but both the table and the graph are based upon consumer reactions which we have observed and cataloged over at least ten years of custom installation work.

"Odd-Room Problems"

Stereo installations would be simple if everyone lived in a perfectly rectangular room, with a wall ratio of approximately 4 X 5, and with no archways, doorways, open windows and broken sections of walls. Unfortunately, architectural aesthetics are often at odds with stereo requirements, which leads us to the bane of the stereo installer's existence—the "L"-shaped living room/dining room combination so prevalent in American architectural schemes.

A typical "problem" layout is shown in Fig. 2. Obviously, the information contained in Table I and Fig. 1 will not work here. The best compromise we have been able to devise for this typical arrangement is shown in Fig. 3. It is based upon the proposition that the extra cubic volume of the dining area is in reality a part of the living room volume (or, at least, about 50% of it is considered to be so) and makes use of the inherently better bass propagating characteristics of at least one of the loudspeakers when it is placed in a corner.

Because bass tones are not particularly directional in stereo programming, this bass enhancement from one corner will not noticeably detract from the overall stereo balance desired. By mentally increasing the dimensions of the major rectangular area by an amount equal to 50% of the dining area, you use the separation figures in Table I and Fig. 1. As an example, suppose the major dimensions (the large area) in Fig. 2 are 15 feet x 20 feet, and the dining area is 10 feet x 10 feet. The dining area is 100 square feet; the main area is 300 square feet. Add 50 square feet (50% of the dining room area) to the major area, for a result of 350 square feet. Keep the ratios of "length" to "width" approximately constant, to arrive at new arbitrary (though fictitious) main area dimensions of 16½ x 22 feet. The nearest available figures in Table I are for a 18 feet x 24 feet listening area, and that is how we arrived at the placement of speakers and listener in the finished layout of Fig. 3.

If you are faced with a situation in which the above calculations indicate that one speaker must be placed in the opening between living and dining areas, the only valid solution (short of compromising the whole installation) is to reverse listening and speaker positions, placing the speakers on the long, continuous wall that is not broken up at all. If such a reversal is necessary, use main area listening dimensions which are **reduced** by 50% of the dining area square footage. In other words, place the speakers **closer together** than would be called for by the basic dimensions of the large main area, per Table I.

Separation Problems Related to Circuit Defects

Of course, things can happen to the electronic components of a stereo system to reduce or completely eliminate separation, and no amount of speaker placement will compensate for these electronic shortcomings, which we shall consider next.

Aside from actual short circuits in amplifiers, the major cause of loss of stereo separation is the inadvertent placement, by the operator, of the "mode" switch in the mono setting.

Similarly, many "one speaker doesn't work" complaints can be traced to complete rotation of the "balance" control either to the extreme left or extreme right position which, in most cases, completely turns off the alternate speaker systems entirely.

In tuners or receivers, loss of stereo FM often can be traced to a selector switch being left in the FM instead of the FM-Stereo (or FM-MPX) position. In such equipment the case is further complicated by the "stereo indicator light", which denotes the reception of a station broadcasting in stereo. Many users erroneously presume that because the light is lit they are, in fact, listening to the program in the stereo mode. Newer sets which feature automatic switching from Mono to Stereo (on FM) have at least eliminated this erroneous use of the selector control; however, even in these sets there is a general "Mono-Stereo" switch which might have been left in the Mono setting.

Phono cartridges themselves can be connected wrongly so that only monophonic sound is produced, even though the stereo cartridge is okay. If the "hot" and "return" leads of one channel of a stereo cartridge are accidentally reversed

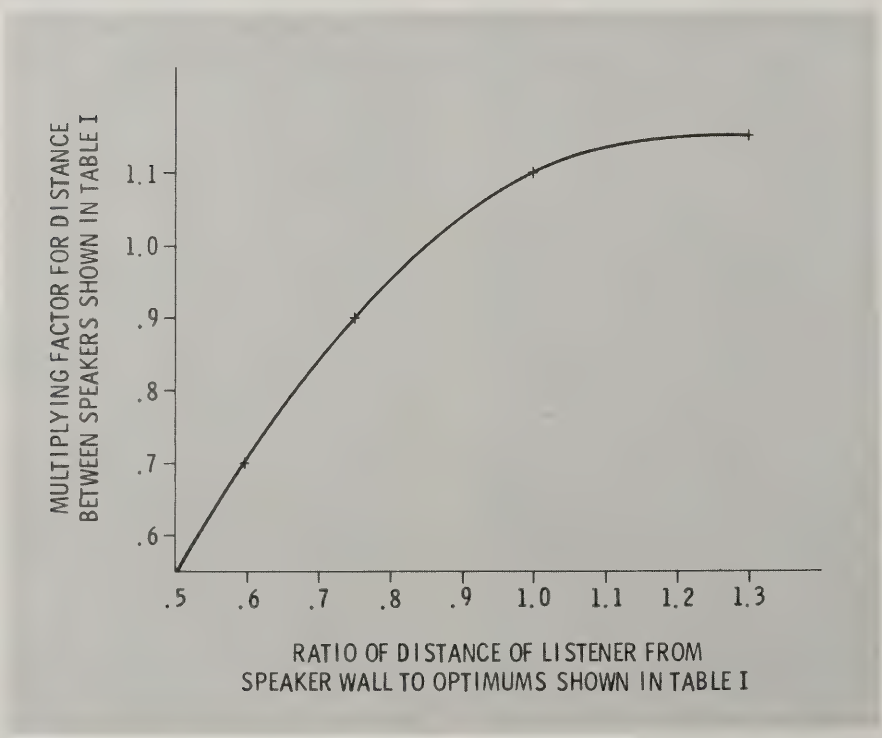


Fig. 1 Graph shows correction factor in speaker spacing when listener's distance to speakers is other than ideal.

while the terminals of the other channel are connected correctly, the only bad result will be phase reversal. If, however, one of the sections of the cartridge is internally grounded, either to the frame of the tone arm or to the frame of the record changer or turntable and this point, in turn, is grounded to the amplifier chassis, a total "short" will have been placed across the incorrectly wired section of the cartridge and sound will come through only one channel.

Many tape recorders also have mono switching arrangements that should be checked first, before checking the circuitry for defects. Remember, in some complex systems a tape deck may be connected to a preamplifier which, in turn, may be connected to an amplifier. All of these elements might have facilities for switching between mono and stereo operation, and if any one of these switches is accidentally left in the mono mode, mono mixing of both channels will take place.

Locating Actual Circuit Defects

The most obvious way to trace stereo separation in amplifiers or preamplifiers is by means of two audio oscillators, each set to a different frequency, each feeding one channel of the amplifier. An oscilloscope can then be used to check for isolation of signals, stage by stage, as shown in the block diagram of Fig. 4. By examining the waveform at each point illustrated

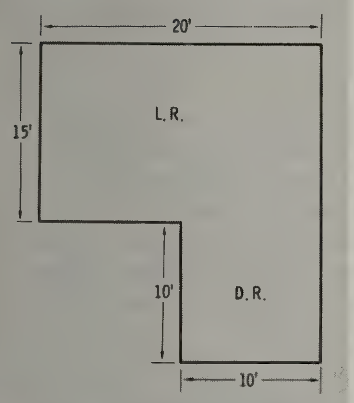


Fig. 2 A typical "problem" layout in stereo installations is the "L"-shaped living room/dining room arrangement.

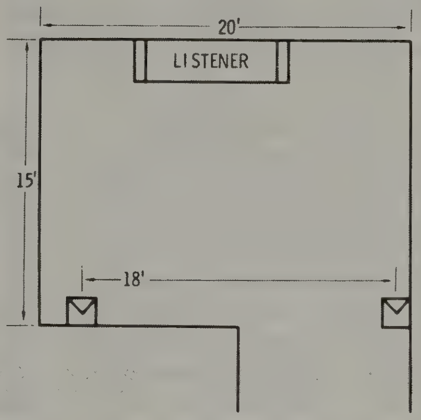


Fig. 3 By allowing for the extra room area, a reasonable speaker arrangement is calculated for the "problem" room of Fig. 2.

Table 1

Speaker placement for best stereo effect

Room Size	Listener's Distance to Speaker Wall	Optimum Distance Between Speakers
8′ x 10′	4′	8′
10′ x 12′	8′	11′ or corners
12′ x 15′	10′	13′
15′ x 20′	12′	15′
18′ x 24′	15′	18′
20′ x 30′	18′	24′

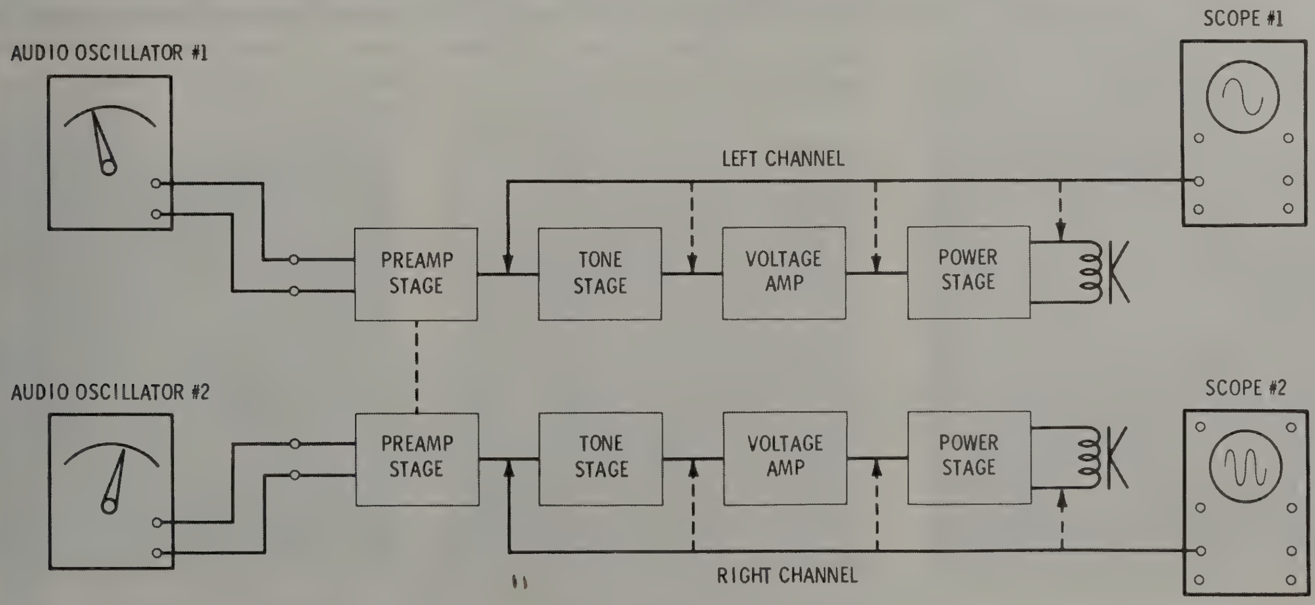


Fig. 4 Separation checks can be made using two audio oscillators set to different frequencies, as shown in the block diagram.

and by using two frequencies which are adequately separated (e.g. 400 Hz for the left channel and 1600 Hz for the right channel), the presence of two frequencies in the channel where only one should be will be very apparent, even if its undesired content is as low as 10% or less (equivalent to separation of 20 dB or more) of the desired signal.

The stage at which such "cross-talk" takes place can then be isolated and further investigated for the presence of shorts between channel components, shorts between "hot" conductors of audio cables from opposite channels, unbypassed power supply dropping resistors, etc., any one of which could be responsible for the presence of an excessive amount of cross-talk.

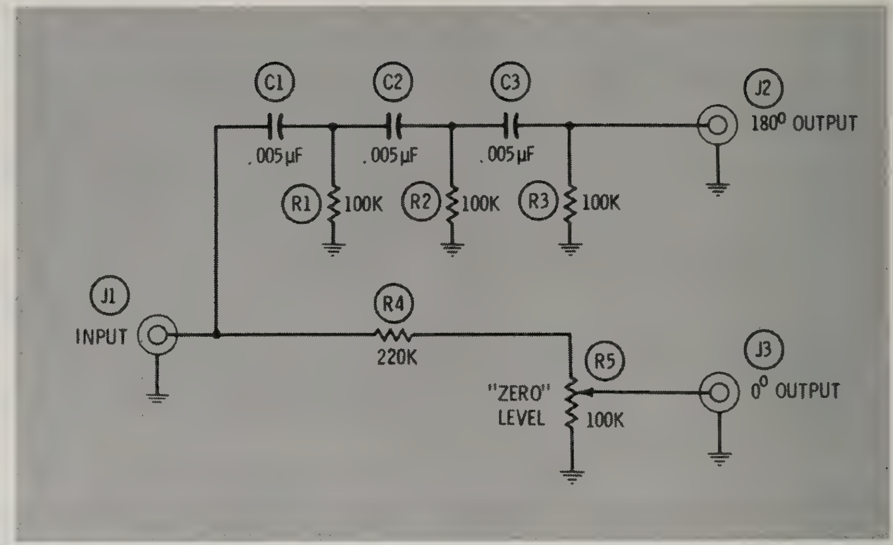


Fig. 5 Schematic of phase-shifting network used to produce two 180-degree, out-of-phase audio signals of approximately 800 Hz.

"Two" Signals for The Price of One

We have devised an inexpensive gadget which we use in trouble-shooting separation problems. This device has two advantages over the method previously described.

Earlier in this article it was stated that the stereo illusion is produced by both amplitude **and** phase differences between channels. The "two-audio-generator" method can tell you all you want to know about the **amplitudes** of the left and right channel signals, but it can't tell you a thing about the **phase** responses.

The device whose schematic diagram is shown in Fig. 5, and which we have labeled, "Two-for-one" adapter, enables a technician to check **both** amplitude and phase of the two stereo channels—and requires only **one** audio generator.

The "Two-for-one" adapter produces two output signals whose phases are 180 degrees apart. As can be seen in Fig. 5, this is accomplished by three RC networks (R1-C1, R2-C2, R3-C3) each of which shifts the phase of the applied sinewave by approximately 60 degrees.

In addition to shifting phase, the networks, as shown, will attenuate the input sine wave by approximately 30 dB. To compensate for this attenuation, a fixed and a variable resistor have been added in the path of the "in-phase" channel. The potentiometer is needed so that the amplitude of the "zero phase" and "180-degree" signals can be set **exactly** equal before the instrument is used for testing. This is a lot less expensive than, say, using 1% resistors and capacitors, which would otherwise have been required to maintain such accuracy. As it is, 10% tolerance for all parts is adequate, and the potentiometer itself can have a 20%, or even a 30%, tolerance.

Since we are using such broad-tolerance capacitors and resistors, the exact frequency at which exactly 180 degrees of phase shift takes place cannot be predicted, other than to state that it will be around 800 Hz (a nice mid-frequency value). This presents no problem, however, because the frequency of your audio oscillator is variable, and you can set its output to the frequency which causes exactly 180 degrees of phase shift in the one channel of the device.

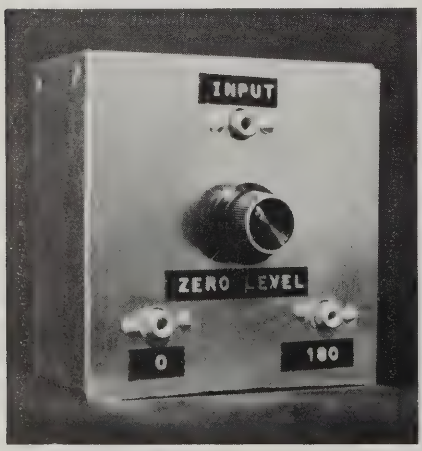


Fig. 6 The completed "two-signal" adapter, labelled for use in tracing separation problems.

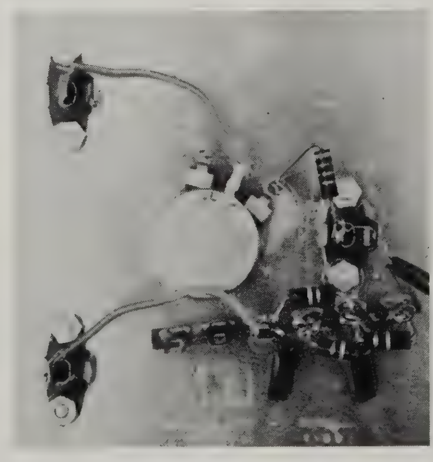


Fig. 7 Rear view of adapter shows wiring layout.

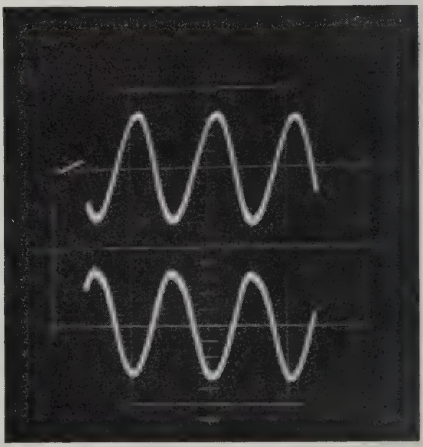


Fig. 8 When properly calibrated, the adapter provides two out-of-phase signals of equal amplitude, as shown in this dual-trace scope photo.

The "Two-for-one" adapter shown in Fig. 6 is constructed on an aluminum chassis, and the inputs and outputs have been labeled for easy identification. The simplicity of wiring is shown in the photo of Fig. 7. Parts layout is not at all critical, because only low audio frequencies are involved. A complete parts list is shown in Table II.

After completing the wiring and assembly, we set out to calibrate the device for use. We had access to a dual-trace oscilloscope and therefore were able to observe the relative phases of the two waveforms as the audio generator was tuned to the correct frequency (in our case, about 780 Hz). The potentiometer was adjusted so that the amplitudes of the two outputs were equal, as shown in Fig. 8.

If you have only a single-trace oscilloscope, connect one of the outputs to the vertical input terminals and the other to the horizontal (external) inputs, and switch the horizontal selector of the scope to "EXTERNAL". If the frequency of the generator is far removed from the frequency which will produce 180 degrees of phase shift a "lissajous" pattern will be produced which looks something like one of the three in Fig. 9. As the generator output approaches the correct frequency, you will first pass through a circular lissajous pattern (indicating 90 degree phase difference) and finally you will see a diagonal line sloping upward, to the left, indicating exactly 180 degrees phase difference between the two signals. Carefully note on your audio oscillator the frequency which produces 180 degrees of phase shift between the two outputs of the device, so that you don't have to repeat this procedure every time you want to use the device for signal tracing.

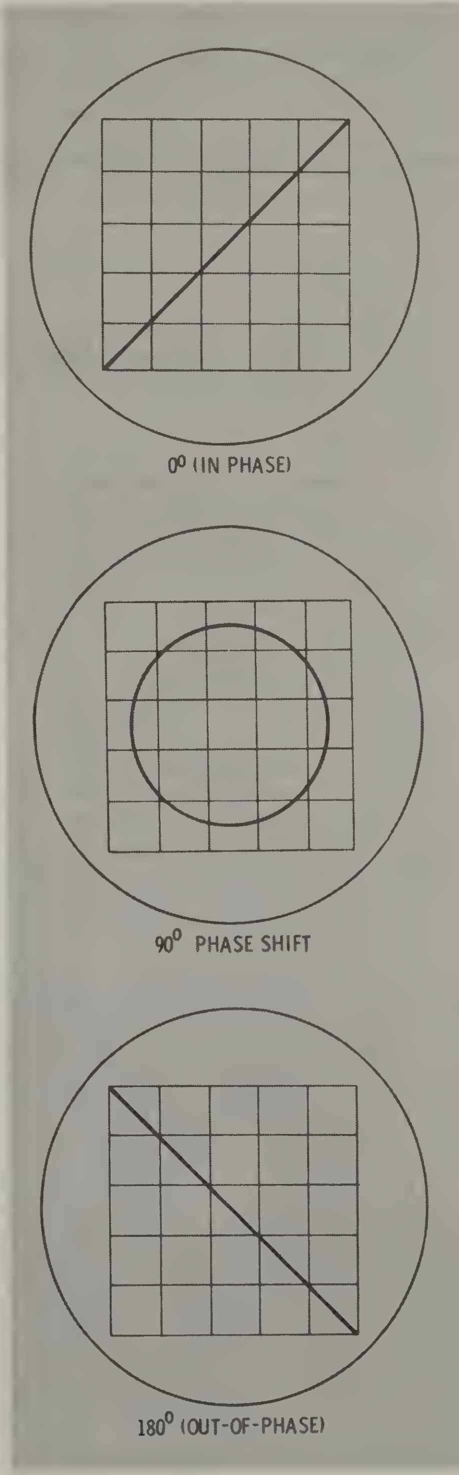
You might have to turn up the amplitude control on your audio oscillator relatively high, to compensate for the approximate 30 dB of attenuation noted earlier. Because most audio oscillators are able to produce 5 to 10 volts of audio output voltage (r.m.s.), there should still be about .15 to 0.3 volt available at the two outputs of the device after it has been calibrated.

The best way to calibrate the **amplitude** of the "zero-phase-shift" signal with respect to the 180-degree signal, using the potentiometer, is to measure each output separately with an AC VTVM, with the correct frequency applied from your audio oscillator.

To trim the frequency and amplitude for optimum, if desired, use a stereo amplifier which is known to be functioning properly and whose separation is at least 30 dB. Feed the "zero-shift" signal to one input, and the 180-degree signal to the other channel. Set the mode switch on the amplifier to MONO and, if the gain of both channels is equal, you should hear practically no sound coming from the speakers. Trim the frequency of your oscillator and the amplitude control on the device for an absolute null (no or minimum sound heard), alternatly adjusting the audio oscillator and the potentiometer on the adapter a bit at a time, until best null is achieved. Your "two-for-one" signal adapter is now ready to troubleshoot stereo amplifiers which exhibit separation problems.

Using The "Two-For-One" Signal Adapter

There are several ways in which you can use this device for tracing separation problems. If the circuit of the defective amplifier is such that mono mixing takes place early in the circuit, you can operate the amplifier in the MONO mode and check for little or no audio voltage between the output points of the respective left and right channel equivalent stages. The stage which first shows a substantial departure from zero audio voltage, as you go down the line, is the one to investigate for possible loss of gain or



0° (IN PHASE)

90° PHASE SHIFT

180° (OUT-OF-PHASE)

Fig. 9 Out-of-phase signals applied to the horizontal and vertical inputs so a scope will produce a diagonal line, as shown here.

Table II

Parts list for phase-shift adapter of Fig. 5

Symbol	Descriptions
C, C2, C3	Capacitor, Disc, Ceramic; 0.005 mfd, 10%, 100V
J1, J2, J3	Jack, Phono, Switchcraft 3501FP, or equal
R1, R2, R3	Resistor, Composition, 100K ohms, ½ Watt, 10%
R4	Resistor, Composition, 220K ohms, ½ Watt, 10%
R5	Potentiometer, 100K ohms, Mallory Type U-41, or equal
———	Chassis, utility, 4" x 4" x 2", Bud Metal AU-1083, or equal
———	Terminal strip, H. H. Smith type 859, or equal
———	Solder, hook-up wire, mounting hardware, control knob

other defect.

If you prefer, you can leave the amplifier in the stereo mode and observe lissajous patterns at the outputs of each stage (of each channel) as before, noting the proper 180-degree phase shift and the equality of amplitude levels at the output of each stage. When a stage fails to meet either condition, the stage should be tested further. Remember that although an extreme difference in phase shift characteristic between stages (observed as a shifting of a lissajous pattern from the desired sloping line, upward to the left), is not strictly speaking, a definite indication of loss of separation or an increase in cross-talk, it nevertheless can impair the stereo illusion and should be analyzed and corrected if possible.

Limitations of the Adapter

It should be realized that this device is useful in only the audio circuits of stereo equipment. Loss of stereo in the stereo FM (multiplex) circuits of a tuner or receiver cannot be analyzed easily using this device, unless the problem is with the one or two audio stages that usually follow the demodulator circuitry. To troubleshoot stereo FM separation problems, a stereo FM generator is still essential. However, the "Two-for-one" adapter, at least, can help you eliminate the audio stages of the system as the source of the separation problem, after which you can proceed directly tc the stereo FM circuitry. ▲

Chapter 5

Isolating the causes of distortion

Harmonic Distortion

The most familiar and common type of distortion in audio systems is harmonic distortion (HD), which is characterized by the undesired presence of frequencies or tones which are an arithmetic multiple of the desired tone or tones. For example, a pure 1000-Hz tone fed into an audio amplifier might produce in the output varying amounts of 2000-Hz, 3000-Hz, 4000-Hz signals, etc. These harmonics change the audible sound produced by the amplifier, and, if present in large amounts, they can change a pleasant sound into a very disturbing one.

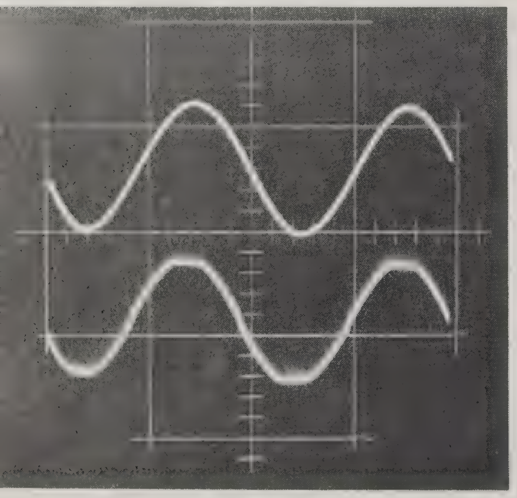


Fig. 1 Dual-trace oscilloscope photo showing undistorted input signal (upper trace) and distorted output signal (lower trace).

It is this "harmonic content" that enables us to distinguish the sound of one musical instrument from another. Middle A on a piano keyboard has exactly the same fundamental frequency as middle A played on a violin, yet we can easily distinguish between the two instruments, primarily because of the percentages of harmonics, or multiples, of the fundamental frequency.

In musical instruments, these harmonics are necessary and desirable. This is not the case in audio equipment. We like to get out of them **only** what we put in—which means that the amplifier should not generate tones of its own.

Fig. 1 is a photo of a scope on the screen of which a double trace has been achieved by means of an electronic switch. The upper trace is a "clean" 1000-Hz input tone fed to an amplifier. The lower trace is the actual output waveform of the amplifier. The "clipped" tops and bottoms of the sine wave represent severe distortion, which would be very apparent to most listeners.

To measure the actual percentage of distortion, the output signal is fed to a distortion analyzer. The distortion analyzer "nulls out" the fundamental, or desired, frequency, leaving only the undesired harmonics, whose amplitudes are measured directly as a percentage of the amplitude of the desired signal. To display the "distortion" on a scope, amplifying circuitry is included in the distortion analyzer.

The display shown in Fig. 2 shows the clean input signal (upper trace) and the distortion product (lower trace). By counting the number of sinusoidal waveforms in the distortion product in Fig. 2, it can be determined that it consists primarily of "second harmonic" frequency, though traces of higher harmonics are present as well.

Experienced listeners can detect harmonic distortion percentage as low as 1%, while less acute listeners can tolerate distortion percentages as high as 5% without noticing them.

The prices of distortion analyzers range from under $100.00 in kit form to about $650.00 for professional laboratory models. This instrument really is indispensable for anyone who services a large amount of high-fidelity stereo component equipment.

Intermodulation Distortion

Another common form of audible distortion which occurs in consumer audio equipment is called intermodulation distortion (IM). Like harmonic distortion, it usually results when an amplifier is driven beyond its power-handling capability. Most experts feel that intermodulation distortion in significant amounts is even more objectionable than harmonic distortion. While harmonic distortion generates undesired frequencies that are directly related mathematically to the fundamental, intermodulation distortion produces totally unrelated frequencies. Certain non-linearities in the amplifier circuitry can actually produce "beat" frequencies in much the same way that a superheterodyne radio receiver produces sum and difference frequencies of the incoming RF signal and the local oscillator. For example, suppose a given instant of musical programming contains a 60-Hz bass tone and a 7000-Hz treble tone. In this case, frequencies of 7060 Hz and 6940 Hz represent the intermodulation distortion products of the amplifier.

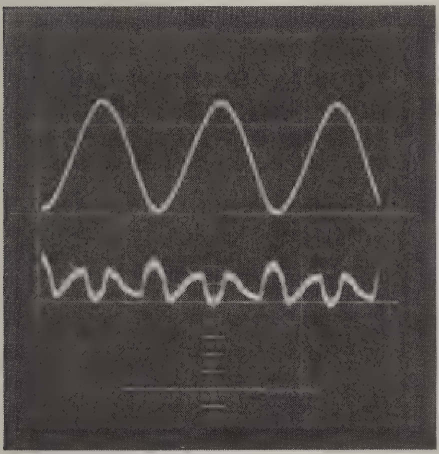


Fig. 2 Distortion analyzer enables the user to analyze the content of distortion. Upper trace is "clean" input; lower trace is amplified representation of harmonic distortion content in output of amplifier whose output waveform is shown in Fig. 1.

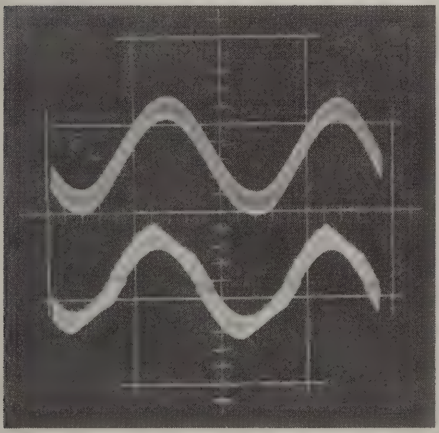


Fig. 3 IM distortion is caused by the non-linear amplification of two widely differing frequencies, which produces sum and difference frequencies. Upper trace is undistorted input. Lower trace is "clipped" output.

This form of distortion also is measured as a percentage of the amplitude of the desired total composite signal and, again, percentages of from 1% to 5% or more can be extremely annoying.

Fig. 3 shows input and output waveforms used in the measurement of intermodulation distortion. The low-frequency component applied is four times the amplitude of the high frequency—a standard agreed upon by the industry. An intermodulation distortion analyzer measures the "beat frequencies" produced by the non-linear overload shown in the lower trace of Fig. 3, and expresses the result in percentage on a meter.

Vacuum tube amplifier designs almost always produce IM distortion that is about four times as great as the harmonic distortion. In solid-state designs, this relationship does not hold true; IM figures generally run about the same or a bit higher than harmonic distortion.

Power output used to be the primary factor related to both harmonic distortion and intermodulation distortion. However, in these days of solid-state amplifier designs that use class B operation, distortion at low power-output levels has become quite prevalent. It is not uncommon to encounter a distortion-versus-power-output curve similar to that shown in Fig. 4, in which the harmonic and/or intermodulation distortion increases not only as maximum power output is approached, but at extremely low power levels as well.

Crossover Distortion

Fig. 5 is a schematic of a simple, low-powered phonograph amplifier using a complementary-symmetry output (push-pull circuit with NPN and PNP transistors fed by a common input). Forward bias for both output transistors is created by the voltage drop between the DC supply and R3 and D1 and R4, and is intended to insure full half-cycle conduction of each of the output devices. If the forward bias is set incorrectly, or changes because of low ambient temperatures or because of a change in supply voltage, etc., each output device will conduct for less than a complete half-cycle, pro-

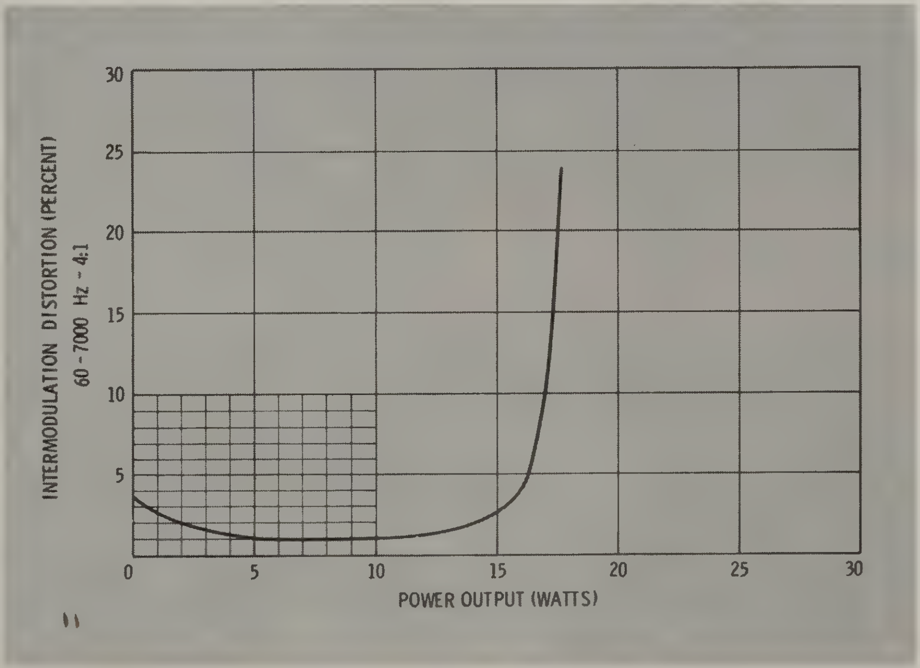


Fig. 4 In some amplifiers, distortion increases with lower power outputs as well as at overload. This conditions usually is caused by "crossover distortion".

ducing an output waveform like that in Fig. 6.

Twice per cycle a brief period exists during which neither output transistor conducts linearly, and crossover distortion is generated. Since the duration of the non-conduction period is constant, it will constitute a greater percentage of distortion at **lower** output levels than it will at higher output levels, and it is for that reason that the overall distortion percentage actually increases as output power is decreased. This form of distortion is particularly objectionable because the harmonic content contributes a "buzzing", almost crackling, effect to the program material.

Amplitude Distortion

Amplitude distortion usually is called "poor frequency response", but in fact it is really as much a form of distortion as the other types already discussed. For example, if the gain of an amplifier is 6 dB higher at 8 KHz than it is at 4 KHz, musical balance will be thrown off if all other elements of the system are flat. Furthermore, if a 4-KHz musical note is being amplified, and if the amplifier contributes 2% of second harmonic distortion (undesired 8-KHz signal), this actually will produce 4% distortion because of the extra treble boost in the amplifier. If this suggests that tone controls should be "banned" from all amplifiers, there is some justification to the premise. Tone controls really should be used only to compensate for other peaks or deficiencies in the response of a system. Unfortunately, most consumers misuse these controls, turning up bass and treble controls to "show off" their systems.

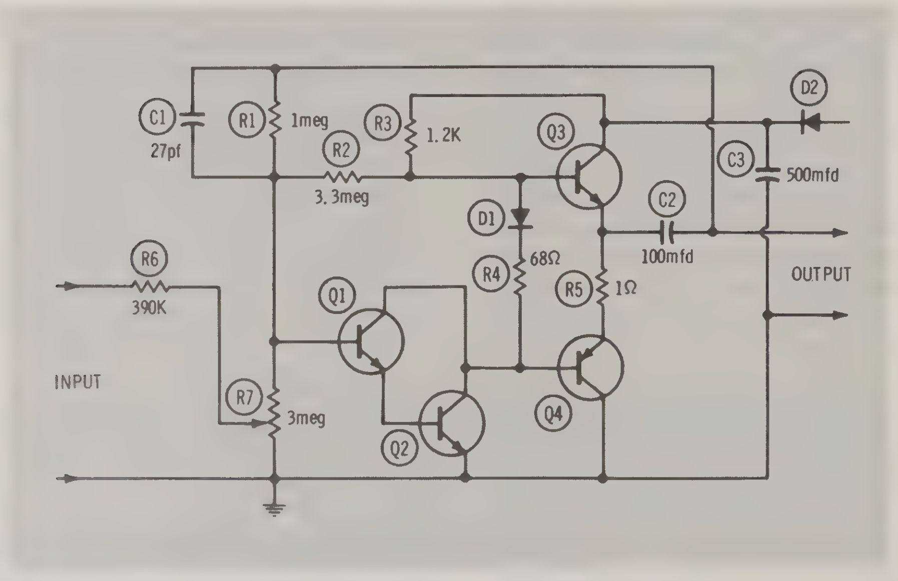


Fig. 5 A typical "complementary symmetry", push-pull amplifier in which forward biasing on output devices must be critically adjusted.

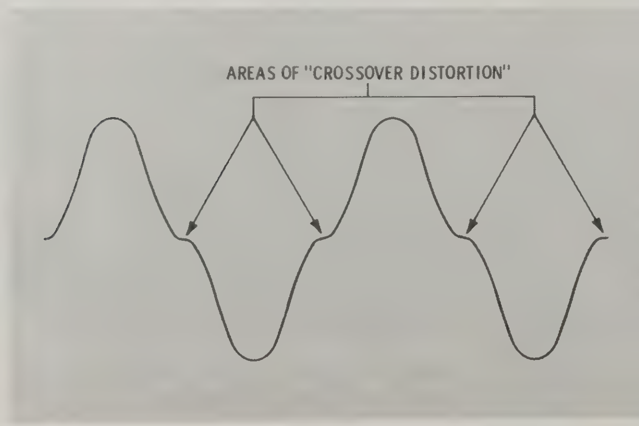


Fig. 6 Output waveform often encountered in class B solid-state amplifiers which have "crossover distortion".

Environmental Distortion

Into this last category of distortion are lumped such miscellaneous conditions as room acoustics, mechanical buzzes caused by "sympathetic" vibration of objects located adjacent to loudspeaker systems, and room resonances which emphasize certain low frequencies out of proportion to others.

Although these conditions are usually relegated to the domain of the professional acoustic engineer, service technicians often can isolate and correct troubles related to these conditions, if they are familiar with them. The addition of draperies or carpeting often can correct an overly shrill-sounding listening environment. Moving stereo speakers to a new location often can "balance", or cancel out, specific low-frequency room resonances. It takes a bit of experimenting and a lot of patience, but it is worth the effort in terms of the customer's satisfaction.

Incorrect Input Levels

The great flexibility of component stereo high fidelity has long been recognized, but it is this same flexibility which often causes high orders of distortion which really are not the fault of any single component in the system.

Fig. 7 is a block diagram of a typical system, consisting of a tuner, a preamplifier, an amplifier and a phono and tape system. The chart indicates minimum and maximum input and output levels for each device. If the system is set up with the actual operating levels shown in Fig. 7, several components will be misused or adjusted to incorrect levels with respect to the mating components. Note that the input stages of the power amplifier will be overloaded rather than the output stages. Under these conditions, there will be audible distortion, even with the amplifier master-volume control turned down to "whisper" levels.

It is of great benefit to have level and volume controls on every component in a system, but it also can lead to severe mismatching of levels if instructions are not followed.

Incorrect Output Stages

Because today's high-fidelity amplifiers are designed to provide fairly high power levels, it is not practical to check maximum power output using loudspeakers, because the sound would be deafening if steady-state single tones were used.

Most high-power wirewound resistors normally used as dummy loads contain a fair amount of inductance, which can upset the output characteristics of some amplifiers, and also lead to erroneous readings.

The best and easiest dummy load to use as a speaker substitute is one consisting of a number of 2-watt carbon composition resistors, arranged in parallel so that their net value equals 8 ohms (the most common high-fidelity loudspeaker impedance). For example, we built two such resistive loads using twenty-eight 220-ohm, 2-watt resistors in parallel for each load. Besides the non-inductive advantages of such an arrangement, this gave us a pair of loads each capable of absorbing 56 watts of audio power **continuously**—suitable for even the highest powered amplifiers.

Two setups for observing output waveforms are shown in Figs. 8A and 8B. In Fig. 8A, observation of the output waveform is made directly by an oscilloscope, and distortion amounts are estimated from experience, in the absence of a distortion analyzer. In Fig. 8B, a distortion analyzer is interposed between the output and the scope, so that distortion content can be observed on the scope as well as measured on a meter. Usually, better amplifiers will have a bias adjustment potentiometer somewhere in the circuit, and if instructions do not detail optimum quiescent output stage current, the best means of adjustment is observation of the scope for lowest distortion as the bias control is rotated from one extreme to the other.

Operating Levels for System in Fig. 7

Component	Input Min.	Input Max.	Output Min.	Output Max.
Tuner	——	——	0 (has volume control)	2 volts rms
Phono Cartridge	——	——	——	0.1 volt rms
Tape Unit	——	——	0.1 volt (has control)	0.5 volt
Preamp: Tuner input	0.1 volt	0.5 volt	0.5 volt	2.5 volts
Phono input	2.0 mv	70 mv	0.5 volt	5.0 volts
Tape input	0.1 volt	0.25 volt	0.5 volt	2.5 volts
Power Amplifier	0.25 volt	0.5 volt	——	50 watts

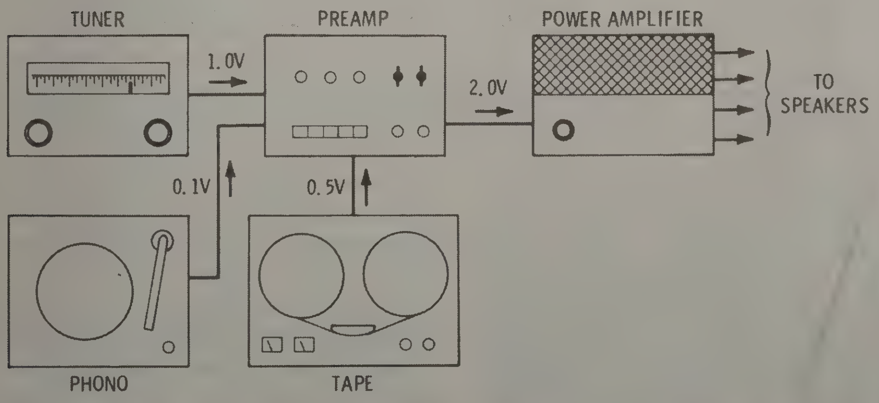


Fig. 7 Typical hi-fi component setup. Accompanying table shows that levels of input to preamplifier and amplifier are misadjusted, causing distortion in first three stages of preamplifier and amplifier.

Program Source Distortion

Besides discrepancies in matching levels between components, distortion often is produced by one or more of the program source elements of a stereo high-fidelity system, such as the FM tuner, the pickup cartridge of the phono system or a tape deck or transport mechanism.

Because many signal sources usually are connected to a system, the first thing to do is eliminate the amplifier as suspect by listening carefully to each of the system's program sources. If distortion occurs with only one of the sources (say, the FM tuner), make sure the output level from this component is compatible with the input-level requirements of the amplifier or preamplifier to which the signal is being fed. If levels seem right, observe the output of the offending system component on a scope, preferably feeding a single mid-frequency tone to the suspect component. In the case of phono or tape, this will mean using a frequency test record or tape. In the case of FM, some form of FM signal generator with built-in modulating frequencies will be required.

Distortion analyzers can be used with "voltage output" components as well as with power amplifiers. Usually, only 0.1 or 0.3 volt is required to calibrate most distortion analyzers to 100% reference points on the meters.

If distortion is noted in the output of an FM tuner, it usually can be traced to poor alignment of the tuner, mistuning or, in rare instances, to the single audio amplification stage often employed just ahead of the tuner output.

Distortion found in phono or tape systems requires a bit more analysis, because these devices include transducers, such as tape heads, phono cartridges, etc., as well as preamplifier stages associated with them. Waveforms should be examined first at the preamplifier outputs of these system components, ahead of the main amplifier. If distortion is noted, a low-level audio signal should be fed to the input of the preamplifier involved. Audio levels for tape head preamplifier inputs are usually no more than 1 millivolt, while for most magnetic cartridges the level varies from under 1

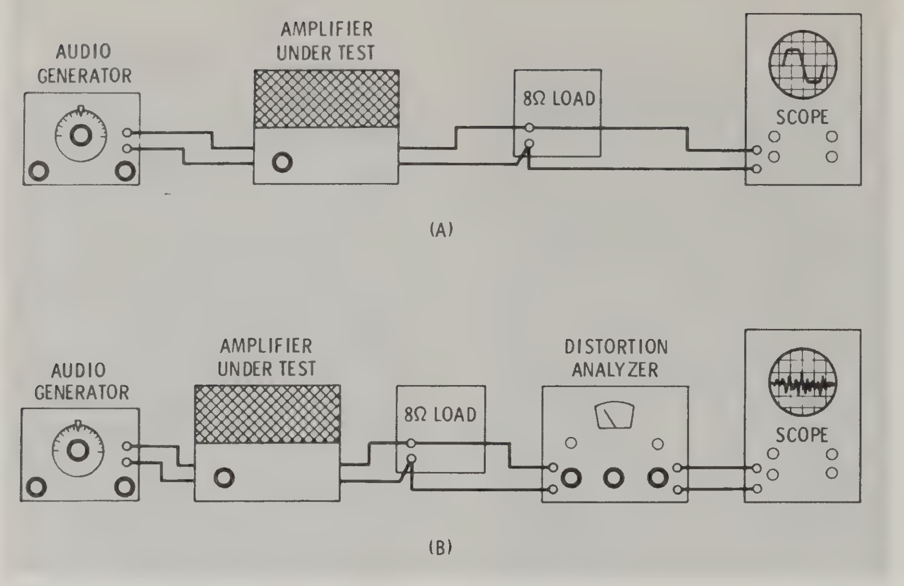


Fig. 8 Two methods used to examine amplifier distortion using: (Top) an oscilloscope connected across dummy output load, and (Bottom) using a distortion analyzer to feed the scope with just the distortion content.

millivolt to about 5 millivolts, at mid frequencies.

If the preamplifier section is distortion-free, the transducer itself should be suspected. In the case of phono cartridges, this can mean a defective stylus which is unable to track the record groove and generates distortion, a mechanically defective coupling between stylus and the rest of the cartridge structure or, in the case of ceramic or crystal cartridges, a defective element.

Don't overlook the fact that the records being played also might be worn, contributing significant amounts of audible distortion. This effect will be first noticed with respect to high frequencies, since these closely spaced undulations in the record groove wall are worn or distorted more easily than low-frequency recorded information. In the case of tape heads, loss of good frequency response, especially high frequencies, means misalignment of the head with respect to the tape travel, or, more often, dirty heads coated with oxide compounds, which are picked up from the tape. Other than these purely mechanical defects, there is little that can go wrong with a tape head, short of an open coil, which would be obvious.

Speaker System Distortion

It is often stated that loudspeaker systems "never wear out". Nothing could be further from the truth. A loudspeaker is called upon to move air by means of a vibrating cone, and anything that vibrates continuously in this manner, and must be suspended and centered with extreme accuracy, is certainly subject to wear.

Often, a rubbing voice coil or a stiff, restricted cone of a loudspeaker can sound very much like a distorted amplifier. The only way to tell the difference is by isolating the two elements by means of waveform analysis.

If you hear distortion in a system, and have eliminated the signal sources as the cause, the next thing to do is to connect your scope at the input terminals of the speaker itself. If the waveform observed there (either by listening to program material or by means of a test tone fed to the amplifier) is still "clean", you can be relatively certain the distortion is originating in the loudspeaker itself.

In a multi-speaker system containing woofer, mid-range and tweeter, placing your ear close to these individual speakers usually will disclose which is defective. Most service shops are not properly equipped to repair speakers which are mechanically defective; this work should be left to the manufacturer of the particular speaker, if possible. ▲

Chapter 6

FM alignment with and without sweep

One of the persistent fears that prevents some technicians from realizing the most from FM radio servicing is alignophobia. This is characterized by a fear of touching tuned circuits above 2 MHz or so.

FM alignment admittedly is more complex than, say, aligning an AM table model radio. On the other hand, even the simplest black-and-white TV alignment is much more complex than FM radio.

Sweep alignment is considered to be the best method for aligning an FM receiver. Although we will shortly be getting into a more simplified method involving non-swept techniques, we'll begin examining the characteristics of FM modulation and sweep alignment methods.

Frequency modulation (FM) differs from amplitude modulation in, shall we say, the direction of the modulation.

AM

In amplitude modulation, the audio information is superimposed on a radio carrier wave. This relationship is shown in Fig. 1. (The waveform shown in C can be viewed on an oscilloscope by connecting the output of an AM signal generator through a diode demodulator probe to the vertical input of the scope.) All that is necessary for proper AM demodulation at the receiver end is a simple solid-state diode detector.

FM

In an FM transmitter, on the other hand, angular modulation is used. There are actually two types of modulation commonly called FM. One is "pure" FM while the other is actually **phase modulation.**

Terminology—deviation and swing

In any given FM modulator, an audio signal of one polarity will cause the carrier frequency to increase while the opposite polarity causes the frequency to decrease. Many technicians become bogged down at this point with unfamiliar terminology and false impressions. One point is confusion over the sweep width and deviation. The distinction between frequency deviation and swing is illustrated in Fig. 2.

The amount of frequency **shift,** usually measured in KHz from the center frequency to either extremity, is called **deviation.**

The entire width of the modulated signal, from it's lowest extremity to the highest, is called the frequency swing. The meaning of this term is synonomous with the sweep width of an FM generator.

Another term, **modulation index,** often is used in connection with frequency modulation. It is defined as the ratio between total deviation and the modulating frequency. Because an FM broadcast station can transmit audio signals up to 15 KHz, the maximum attainable modulation index value is 5 (75 KHz deviation/15 KHz modulating frequency).

Some technicians believe that deviation and frequency swing are determined by the frequency of the modulating signal. This is true for **AM.** In pure FM, however, deviation and frequency swing are determined by the **relative level,** or amplitude, of the modulating signal. The term "100 percent frequency modulation" is actually an arbitrary standard established by the FCC, who have determined that 75 KHz deviation will be 100 percent.

Phase modulation

To prevent confusion, it might be well to explain a thing or two about a related form of angular modulation, phase modulation (PM). It is the type of modulation used in the so-called VHF-FM two-way radio transmitters.

PM deviation is directly related to the modulating frequency. The process of phase modulation produces a rising curve which increases 6 dB per octave over the range of modulating frequencies. Because of this, it is necessary to **de**-emphasize the higher audio frequencies.

The deviation vs modulating frequency differences between FM and PM are shown in Fig. 2B.

FM fidelity

Frequency modulation is well known for it's high-fidelity sound. This is attributable to the wide range of audio frequencies allowed FM broadcasters by the FCC. An FM station is allowed to transmit audio frequencies up to 15 KHz while most AM stations are restricted to no more than 5 KHz. Some hi-fi buffs might be more than a little upset if they knew that FM is actually rather lo-fi. Those upper audio frequencies must be increased, or as mentioned earlier, pre-emphasized, before being applied to the modulator in the transmitter. To compensate for this pre-emphasis, the receiver must be equipped with a 75-microsecond (RC time constant) de-emphasis network.

The purpose of sweep alignment is to insure that the bandpass of the receiver is wide enough to provide sufficient amplification of all the transmitted intelligence. An FM receiver that has too narrow a bandpass will significantly distort the received signal. On the other hand, a bandpass that is too wide will produce an annoyingly high noise level.

What constitutes proper receiver bandwidth seems to be a matter of controversy. Many technicians, including the author, have traditionally aligned FM receivers to produce a 150-KHz bandpass. However, at least one authority claims that it should be wide enough to admit the eighth significant harmonic produced by an index 5 modulating frequency, which is 240 KHz on monoaural stations and 318 KHz on stereo stations. Because there is contradiction about this point, and because an electronic technician is a professional who is responsible for his own work, I leave it to you to determine which bandwidth to use.

Sweep Alignment

A typical sweep alignment setup for FM is shown in Fig. 3. The

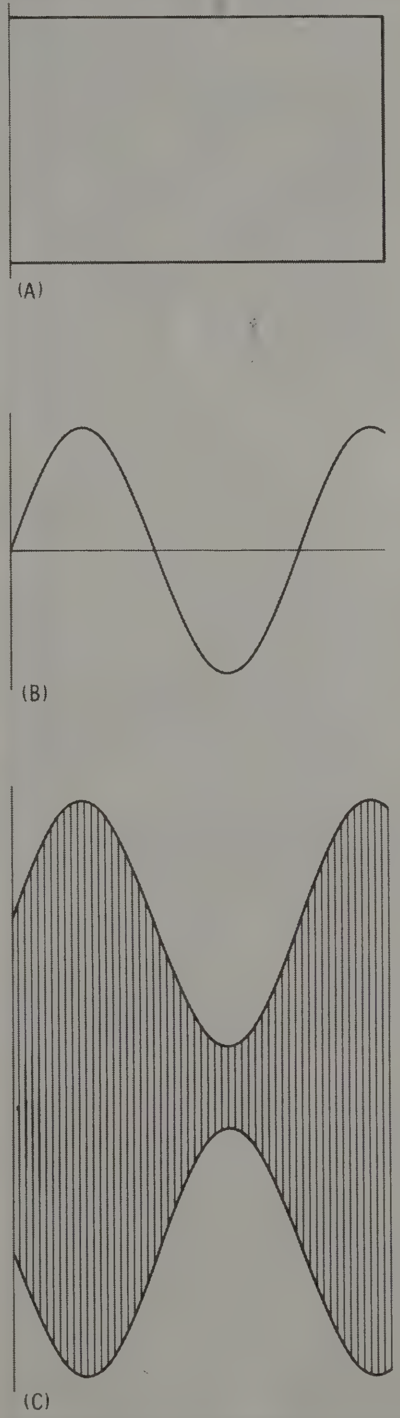


Fig. 1 Elements of amplitude modulation. **A)** Unmodulated carrier. **B)** Modulating signal. **C)** Amplitude modulated carrier. All of these can be viewed on a scope.

sweep generator simulates the signal from an FM broadcast station. The marker generator provides small pips which help determine the frequency at a particular point on the response curve traced out by the oscilloscope. The marker and the sweep outputs are fed into an adder. This device combines all of the required signals into one composite that can be fed into the FM receiver. Direct connection, without the adder, can cause interaction between sweep and markers which can distort the response curve. Many test equipment manufacturers now offer sweep, marker and adder functions in one cabinet. Many of these instruments also include bias supplies and other functions that are needed by TV men. All this means a lot less effort, less bench clutter and more usable bench space.

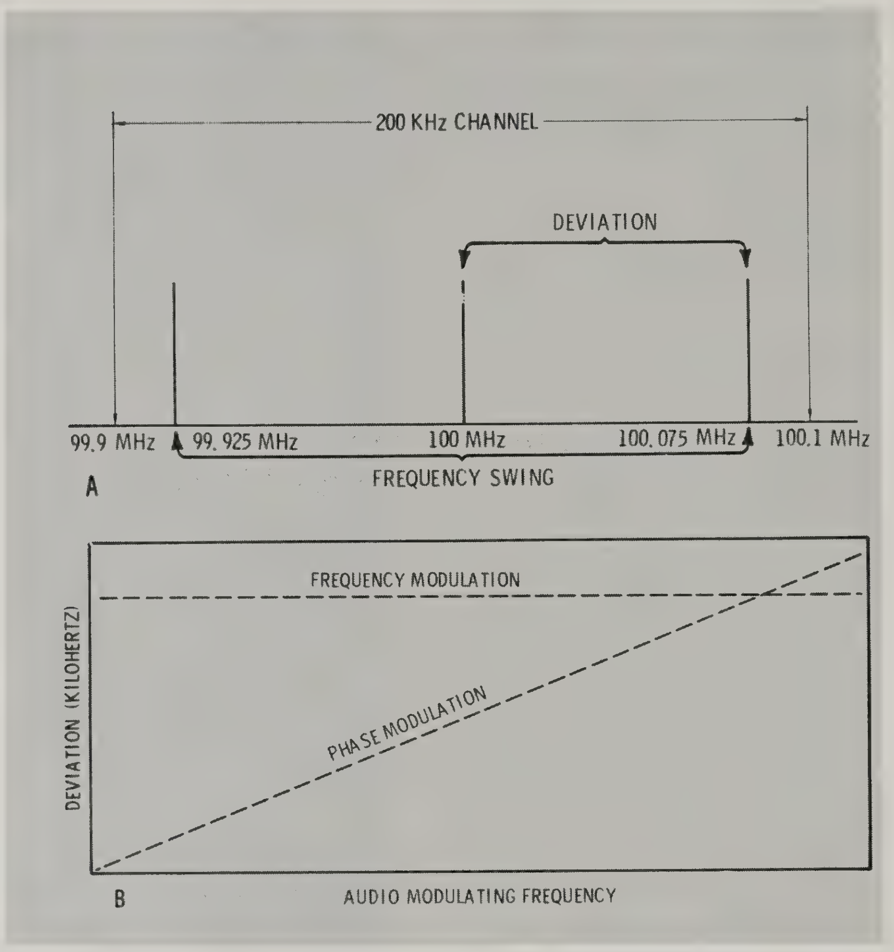


Fig. 2 A) The graph here illustrates the frequency components of a single FM channel. The total bandwidth is 200 KHz, 50 KHz of which is divided into two 25K-Hz guard bands, one situated at each extremity of the channel. The amplitude and polarity of the modulating signal causes the frequency to shift either above or below the center, or carrier, frequency; this is called deviation, and is limited to no more than 75 KHz above or below the center frequency. The total amount that the modulating signal can shift the frequency is 150 KHz. The frequency of the modulating signal determines the rate at which the frequency of the transmitted signal is shifted. Ideally, the amplitude of the transmitted signal remains relatively constant. The percentage of FM modulation, usually called the modulation index, is determined by the ratio of the maximum deviation and the modulating frequency which produced it. For example, if a deviation of 75 KHz is produced by a 15KHz modulating signal, the modulation index is 5, or 100 percent—the deviation and the modulating frequencies used in this example are the maximums allowed by the FCC. **B)** This graph illustrates the fact that in phase modulation, which is an indirect form of frequency modulation, the higher frequencies of the modulating signal are purposely de-emphasized prior to application to the modulator. This is because the amount of frequency modulation produced by the lower modulating frequencies is not proportional to that produced by the higher frequencies; the lower the frequency, the disproportionately lower the modulation. The main advantage of the various phase-modulation systems over conventional FM systems is that a crystal-controlled oscillator can be used in the master oscillator. This eliminates the need for the separate crystal-controlled frequency control system required to maintain acceptable frequency accuracy in conventional FM systems.

IF

The alignment instructions for most FM receivers and/or tuners usually will tell you where the adder output is to be injected. Success will be more sure if these instructions are followed to the letter.

If such instructions are not given, however, you can connect the output wire from the generator or adder to a short piece of insulated hook-up wire. This "gimmick" is then dropped inside the slug of the 1st FM IF transformer. If this transformer has screw driver slots instead of hex-holes in the slugs, the added output can be connected to the base or grid of the FM mixer. The hot side of the adder's output cable should be connected to the injection point via a capacitor of .001-.005mfd.

Most instructions also specify the generator settings. These typically read: 10.7 MHz, 22.5 KHz sweep width, 400 Hz modulation.

A simplified block diagram of an FM receiver is shown in Fig. 4. Connect a zero-center, high-impedance voltmeter to point A. (Most ordinary shop VTVM's can be made zero center by readjusting the zero control.) With proper signal applied to the input point, as described previously, adjust the secondary of the detector transformer for zero volts. The meter will read a positive voltage on one side of the correct transformer setting and a negative voltage on the other side.

Next, connect the meter across the speaker of the receiver. Peak all other IF tuned circuits to produce a response curve similar to that in Fig. 5.

Tuner

To align the front end, it is necessary to apply a proper signal to the antenna terminals of the set. The signal, preferably, should have a frequency that corresponds to one of the calibration points on the tuning dial.

After injecting the proper signal, zero the receiver oscillator by carefully turning the associated trimmer capacitor (occasionally a slug-tuned coil is used) until the voltage at point A (Fig. 4) is again zero.

Next, peak the RF amplifier and antenna tuning adjustments for maximum output. Monitor the response curve on the scope during this process. The markers will help you determine whether or not the response is correct at the significant frequencies. It is best to sacrifice a little gain to produce a correctly shaped response curve.

Non-Swept Alignment

Although sweep alignment is considered the most accurate way to align an FM set, it isn't the only way. An unmodulated signal generator, if properly used, can provide a signal good enough for acceptable alignment under certain conditions.

Equipment for non-swept alignment is wide and varied. In general, however, all of it can be broken down into two catagories: First, you will need some sort of level indicator. A good shop VTVM generally will suffice. Also, you will need a stable signal source. This can be either a good grade of tunable service-type signal generator

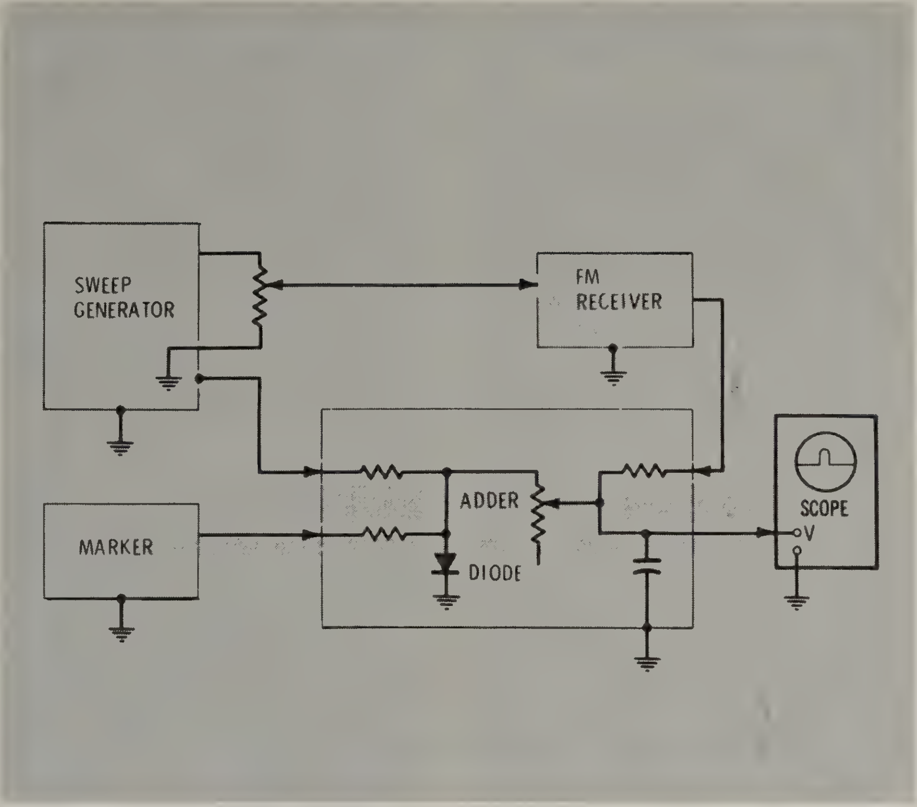


Fig. 3 Typical setups for performing sweep alignment of an FM receiver. See text for detailed procedure.

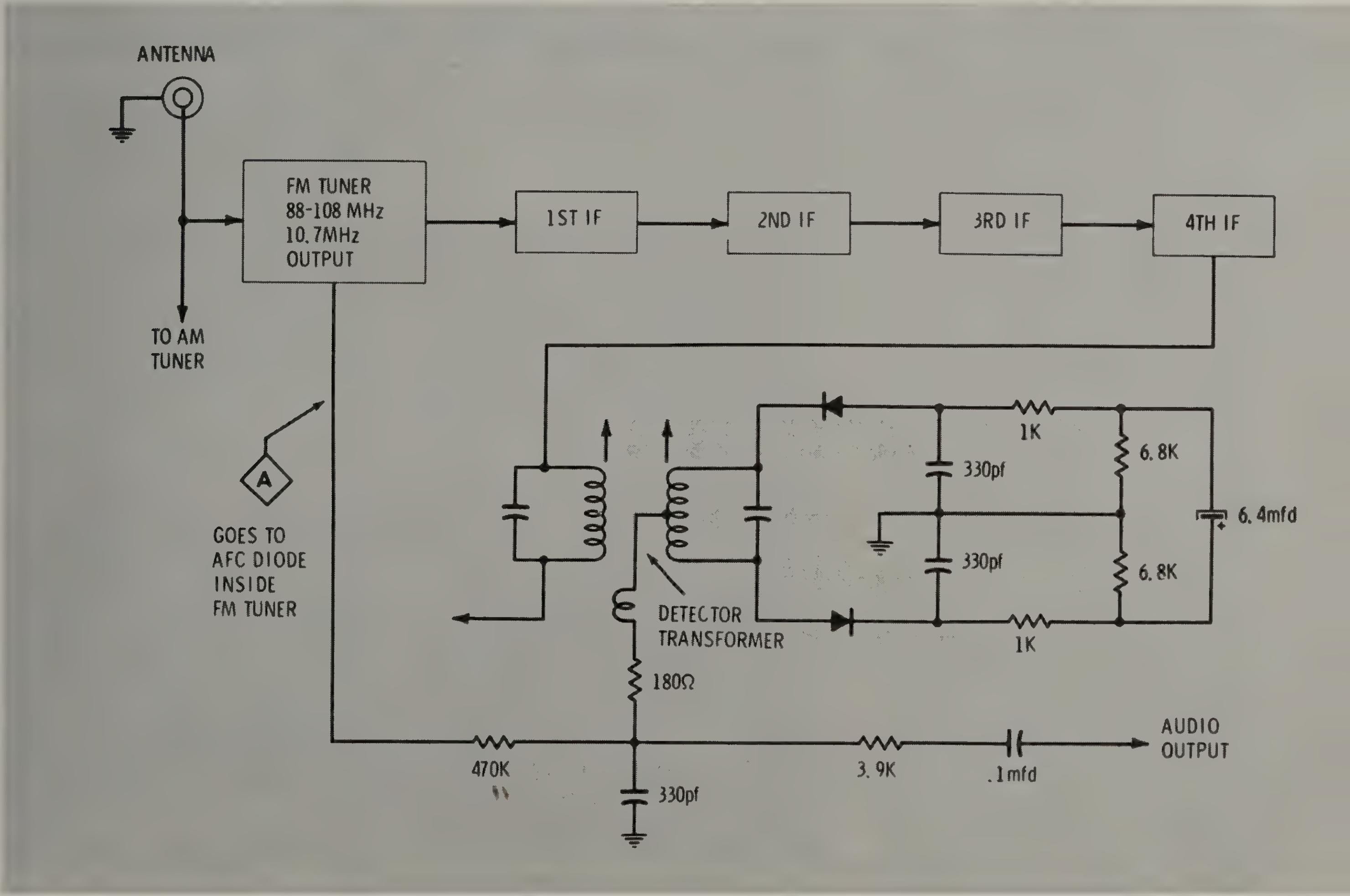


Fig. 4 Simplified block diagram of a typical FM receiver (Motorola Model FM991).

(AM modulation, if any, must be switched off) or a crystal-controlled marker oscillator.

There are only a few requirements which the signal source must meet: One is that it have decent short term stability. Another is that it be reasonably accurate. A third requirement is that it have low leakage and a good attenuator. (Many of the so-called service grade generators exhibit enough leakage, even under maximum attenuation, to cause overload problems during alignment.)

The author built a fairly decent alignment oscillator using commercially available oscillator and buffer kits. These are inexpensive and, when housed in a double-shielded aluminum enclosure produce low residual RF radiation.

I chose two crystals for my portable FM alignment oscillator: 10.7 MHz and 9 MHz. The 10.7-MHz crystal is, of course, used to align the FM IF amplifiers. The 9-MHz crystal was chosen because it has harmonics at 90 MHz, 99 MHz, and 108 MHz, which can be used for checking an FM receiver's calibration and tracking at both extremities and the middle of the band. No provision was made for modulation because this signal generator was intended exclusively for FM alignment and servicing.

Many sets require some sort of dummy antenna between the signal generator and the injection point. This is especially true of automotive FM and FM Stereo receivers. A dummy antenna which can be used with most types of car radio is shown in Fig. 6. It can be built inside of a standard Motorola-type car radio antenna plug. Other types might require a different type of dummy antenna. Be sure to consult either the appropriate PHOTOFACT or the manufacturer's service manual.

The test setup shown in Fig. 7 will produce satisfactory alignment of most types of FM receivers. Connect the output of a 10.7-MHz generator (6.5-MHz for some European car radios) either to the input of the FM mixer or, via a gimmick, as described earlier, to the 1st IF transformer. (Do not connect the generator cable directly to the transformer because the cable capacitance will de-tune the set.) Connect a high-impedance DC voltmeter to a point that corresponds to point Z in Fig. 8. Adjust the secondary of the detector transformer to null (zero voltage). Here a zero-center VTVM will be convenient. As in sweep alignment, the voltage will swing positive and negative on either side of the correct adjustment.

There are two major factors which, if overlooked, can make identification of the "null" point difficult. One is overloading of the IF amplifier chain, which, in some sets, is caused by AGC action. Identification of the null points also can be difficult because the visual effect of the tuning adjustments will be very small in relation to the total meter reading.

The AGC system can be disabled in many FM sets. In fact, in many cases, the manufacturers instructions call for the FM AGC to be disabled.

The only way to eliminate simple overload problems associated with alignment is to select more attenuation in the signal generator. Most instructions state that the best signal level is the point slightly above the level that "quiets" the FM receiver. As the tuned circuits are peaked up, it might be necessary to periodically readjust the attenuator to that point.

IF alignment

To peak the IF transformers, connect the meter to the point that corresponds to Z in Fig. 8. An alternate point might be the input of the limiter stage, which will require a low-capacitance detector probe. At either test point, the reading will be maximum when the IF transformers are properly adjusted.

When using the detector probe at the limiter input, however, you will find that adjusting the limiter output tuning will not produce an indication on the meter. Consequently, to peak the limiter output, connect the meter to either side of the detector transformer secondary. Be sure to use only a high-impedance voltmeter equipped with detector probe.

During IF transformer peaking, the signal source may be left connected in the same manner as it was in the previous step. Again, be sure that the generator output level is maintained just above the receiver's quieting point, to prevent overload.

Do not attempt to use a broadcast station signal as a source. It is very difficult to find a station that presents this optimum signal level with any degree of reliability. Also to be considered is that "capture effect" and the inherent rejection of weak signals by FM receivers will raise havoc with attempts at on-the-air alignment.

Also, do not attempt to use your ears as a level detector. Added to the objections listed previously is the annoying broadness of FM tuned circuits. This broadness is a result of the 75-KHz maximum de-

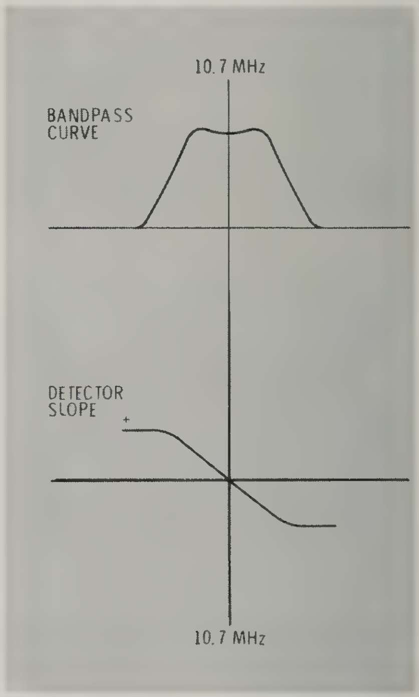


Fig. 5 Typical response curves produced by properly tuned FM receiver.

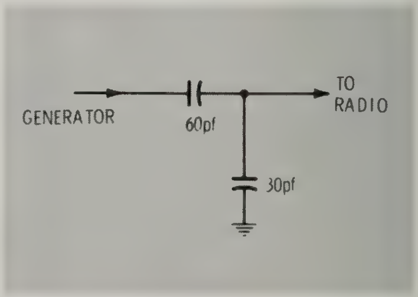


Fig. 6 Circuitry of a dummy antenna which can be used between the signal generator output and the signal injection point of most auto FM receivers.

viation. Also, the receiver can break out of quieting at both extremes of the tuning slug adjustments while the quieted range can cover several complete turns. This explains the futility of aligning by ear.

Work from the detector end of the chain towards the tuner. Adjust first the secondaries then the primaries of each transformer. When you have completed the sequence, start again and repeak for optimum results. This repeaking is required on many sets because the transformers have substantial interaction.

Tuner alignment

The alignment of the tuner section of the FM set is similar to that used in sweep alignment. However, many of the signal sources that were usable at 10.7 MHz are useless at 100 MHz.

Some RF generators use harmonics to supply signals over 30 or 40 MHz. Unfortunately, this leads to problems. A frequency error, for example, at 10.7 MHz might be acceptable. At the tenth harmonic, however, that error will be ten times as bad. Instability also will be ten times as bad. Therefore, a signal generator that had reasonable short-term stability at 10.7 MHz will be unacceptable at 100 MHz.

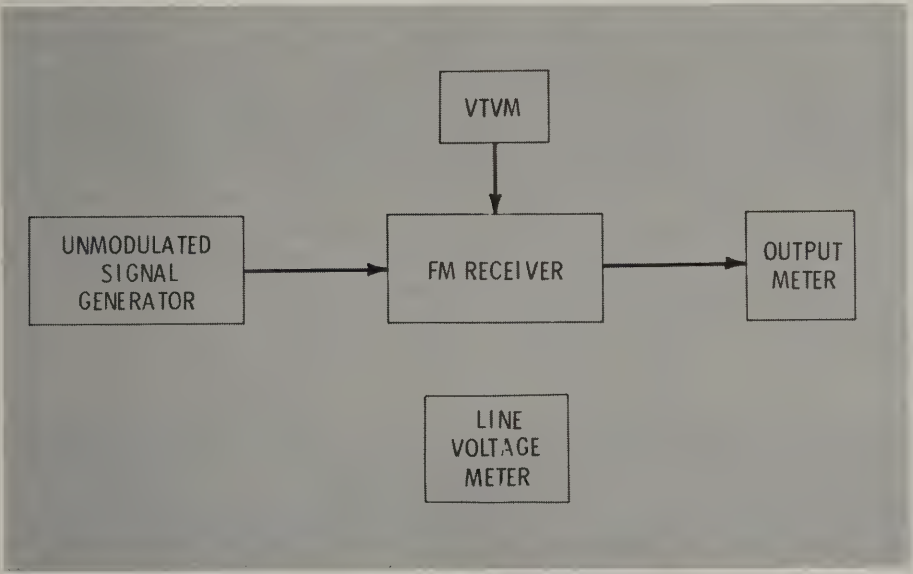


Fig. 7 Equipment setup for performing non-swept alignment of most types of FM receivers. See text for detailed procedure.

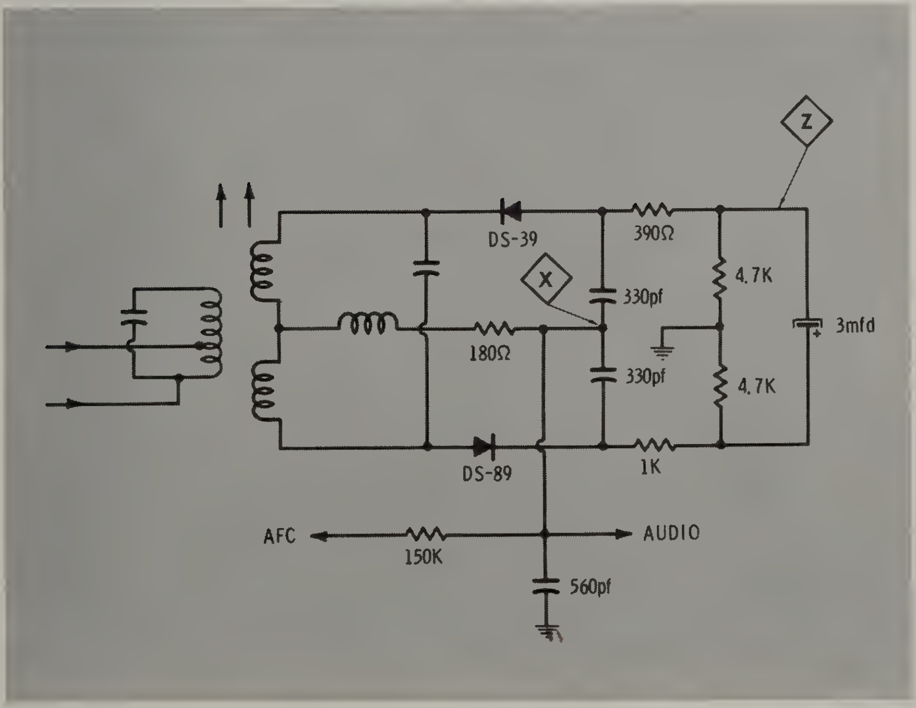


Fig. 8 A high-impedance DC voltmeter, connected to point Z in the demodulator circuitry shown here, serves as the indicator during non-swept alignment.

As an alternative signal source, I use the 9-MHz crystal oscillator described earlier. Another alternative is to use the second harmonic of a crystal controlled TV alignment marker generator. The 2nd harmonic of the 44-MHz marker, for example, is 88 MHz. The harmonics of other markers provided other required frequencies further up the FM band. Again, as with the IF stages, continually adjust the attenuator in the generator to just above the receiver quieting level.

The tuner of an FM receiver is especially sensitive to changes of stray inductance and capacitance. To combat this problem, be sure to use a rather long non-metallic alignment tool. Also, be sure not to move any leads, resistors, capacitors, or coils inside the tuner. Although it might be a great temptation to move a disc ceramic capacitor to get to a tuning adjustment, it can very easily upset prior adjustments. Therefore, make sure all adjustments and test points are accessible before beginning alignment.

Proceed very slowly when adjusting the tuned circuits in an FM set; you might miss the peak setting of the adjustment.

It is worth noting, at this point, that most trimmers used in FM tuners pass through their entire capacitance range in just 180 degrees of rotation, although they will rotate a full 380 degrees. This will produce a double peak that can be very confusing.

Another standard FM alignment procedure is to leave all metal covers in place during alignment. It is quite possible for these covers, which admittedly are difficult to work through, to completely change the tuning of the front end.

Quadrature techniques

The methods detailed previously

have been predicated on the assumption that the set uses either a discriminator or ratio detector. For most sets, these techniques will easily help a technician through an alignment job. Some recent sets, however, do not use either of these types of detector. They are using what is known as the quadrature detector.

In most sets employing quadrature detection, the demodulation, limiting and final IF amplifier functions are combined inside one integrated-circuit chip. This chip can be either a 14-lead dual in-line IC pack or a multi-lead version of the old TO-5 transistor case. In any event, quadrature detectors require an entirely different alignment problem.

The circuit in Fig. 9 is the quadrature detector circuit used in the 1970 Delco AM/FM Stereo radio model 02GFP1 (installed in Pontiac Grand Prixs). The alignment procedure for this radio is different than that for conventional detectors. One apparent difference is the necessity for using an RF demodulator probe as a peak indicator. Delco recommends the RCA WG-301 demodulator probe be connected to a high-impedance VTVM.

A 10.7-MHz unmodulated signal is injected into the set at the base of the mixer transistor. On the Delco set, it is necessary to use a dummy antenna consisting of a .0047-mfd. capacitor in series with a 270-ohm resistor.

The RF probe is connected to point M in Fig. 9. Adjust the transformers between the first two stages for maximum amplitude.

First, to adjust the phase transformer in the quadrature detector, remove the RF probe from the VTVM. Switch the meter to "AC Volts", then connect it directly across the speaker terminals. The 10.7-MHz signal is injected directly into the antenna circuit of the receiver. Reduce the output of the generator until background noise begins to overcome quieting. As the phase transformer is adjusted through its entire range, you will notice there are two peaks in the background noise. The proper adjustment point is the null between those two peaks. It will be very close to midway between the peaks.

Although the author prefers to use a genuine FM sweep generator for FM alignment, non-swept methods do have their proper use. After replacing a transistor or FM tuning component, for example, it is generally only necessary to peak the affected components using the above techniques. About the only place we use non-swept techniques for a complete alignment is when it is on a low-cost, low-quality set whose performance will not be compromised by such an alignment. ▲

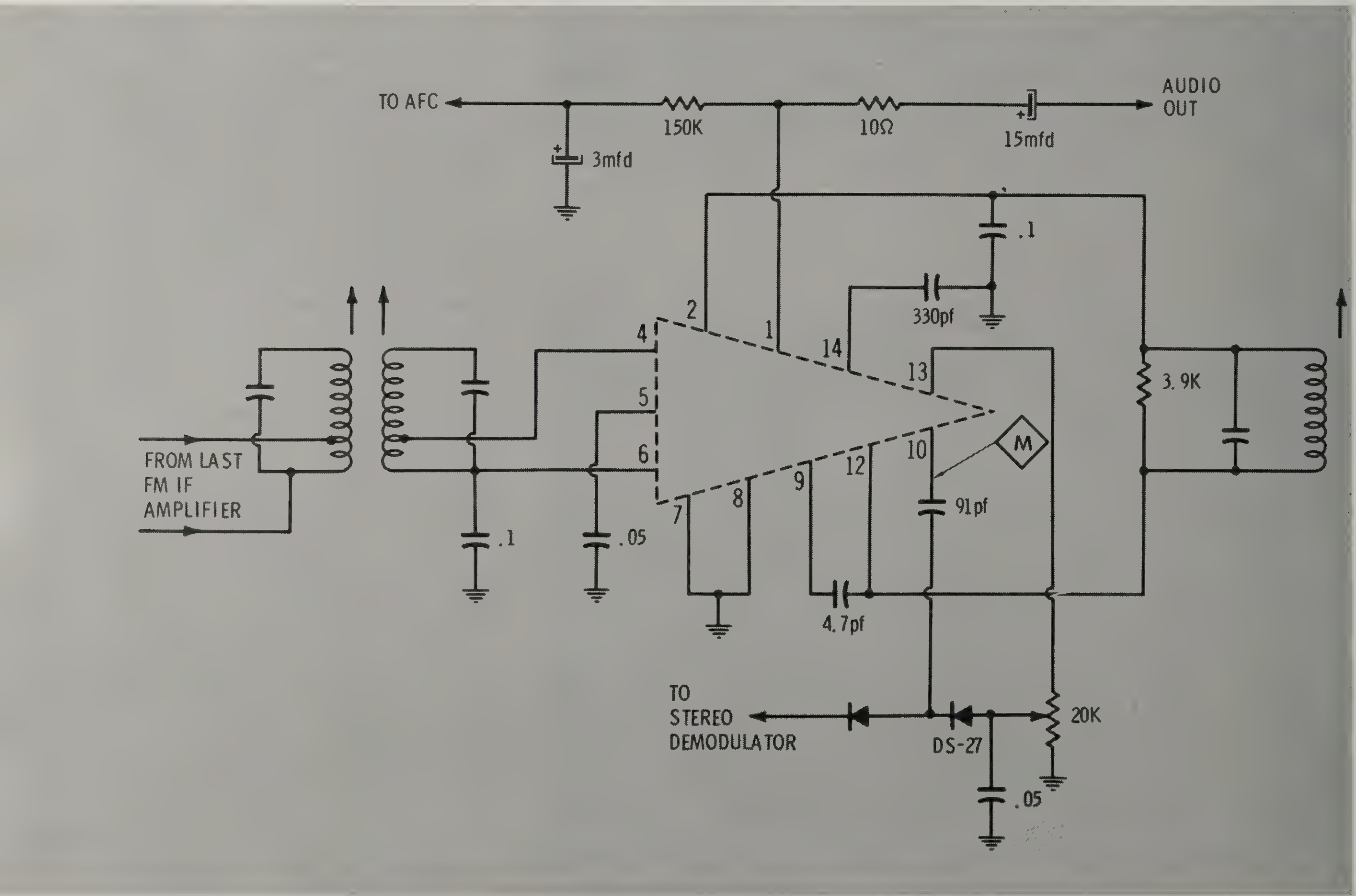


Fig. 9 Quadrature detector circuits, such as this one, used in Delco AM/FM stereo radio Model 02GFP1, require an alignment procedure which is different than that used with receivers equipped with conventional detectors. See text for detailed procedure.

Chapter 7

Stereo generator selection and application

Facts you should consider when selecting a stereo generator, and, after you've made your choice, how to properly use it to align and troubleshoot FM stereo systems.

If you've tried to align stereo FM circuits using "off-the-air" signals, you've probably decided it can't be done. Circuits are entirely too critical, and, unlike the early days of stereo FM, broadcast stations today seldom provide left-only or right-only signals which you can use as a "test signal". For proper alignment of stereo FM receivers, a stereo FM signal generator is a must.

Before you decide which of the many stereo generators will best serve your needs, there are a few things you should know about the types available. **After** you've decided on the make and model, there are a few more things you should know to make your investment pay off. Both the before- and after-purchase tips are the subjects of this discussion.

Selecting One of Two Basic Types of Generators

The earliest stereo FM generators offered to service technicians and engineers usually provided a composite stereo signal only. The single signal available at the output of these generators included the right proportions of 19-KHz pilot carrier, a left-only or a right-only audio signal (internally supplied by means of an audio oscillator with one or more selectable frequencies), and, sometimes, a means for externally modulating the composite signal by means of a separate audio oscillator.

Later, manufacturers combined the composite signal generator with built-in RF oscillators which could be FM-modulated by the composite signal. Usually, the RF oscillator section provided a non-attenuatable RF output signal of several thousand microvolts.

Such a high-amplitude RF signal is fine for making quick checks of the overall performance of a stereo FM tuner or receiver, but is not suitable for stereo sensitivity measurements or any "weak signal" alignment work such as would be necessary when working on quality, high-fidelity equipment. Also, the RF frequency of the all-in-one generator usually is pre-set to about 100 MHz, with a fine-tuning control which permits changing frequency only a few hundred KHz in either direction. Unless you are in a very poor FM reception area or have a shop equipped with an expensive "screen room", the middle of the FM frequency band in your area probably is loaded with station signals which would tend to interfere with your measurements.

If you already own a decent FM generator **capable of being externally modulated** by frequencies up to 67 KHz, purchase a stereo generator which provides **only** the stereo composite signal. If you don't own an FM generator, choose a stereo FM generator which produces its own RF signal.

If you have a good FM generator, first determine how many volts (rms) it takes to externally modulate it to full 75 KHz deviation. Some generators require only one or two volts of external audio for full external modulation, while others require as much as 10 volts rms. Make certain that the stereo generator you buy provides sufficient amplitude of composite stereo signal to fully modulate your FM generator.

Another important point to consider is the **separation capability** of the generator. Some inexpensive stereo generators provide a maximum separation capability of little more than the minimum 30 dB required by the FCC for all stations transmitting stereo programming. Some of the better stereo tuners and receivers boast separation capabilities of 40, and even 45, dB—a claim you will never be able to veritfy with a generator that puts out a signal with inherent separation no better than 30 or 35 dB.

Finally, you must decide how important it is for you to be able to check separation at **all** frequencies (and even demonstrate stereo programming to your customers using your stereo generator). While some generators have enough built-in audio frequencies to check separation at low, middle and high ends of the audio band, not all have additional provisions for external application of left and right programming of your choice.

Checking Out the Generator

Regardless of which type generator you select, the first thing you should do is verify its rated performance. Using a stereo tuner or receiver for this purpose is rather meaningless, unless you are absolutely certain of the specifications and state of alignment of the tuner or receiver. Instead, make a few

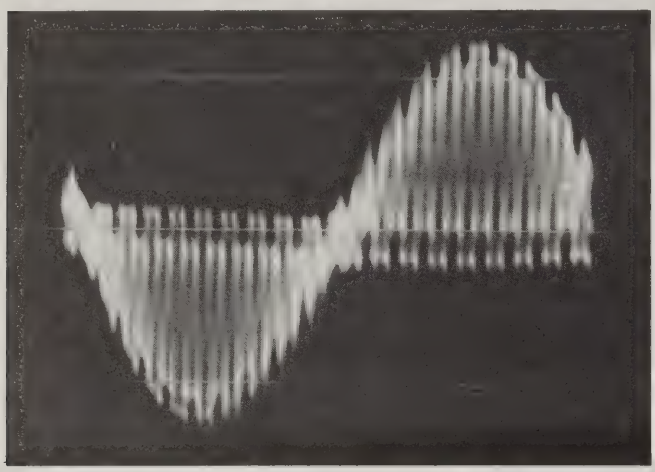


Fig. 1 Scope photo shows one complete cycle of composite stereo signal, with left-only (or right-only) 400-Hz audio signal applied.

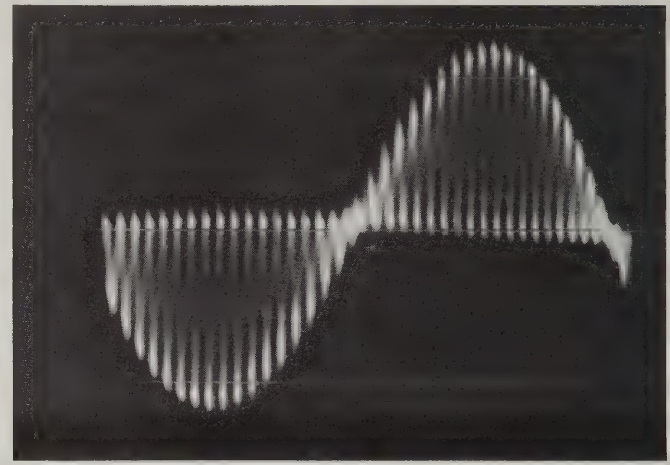


Fig. 2 Composite stereo signal of Fig. 1, with 19-KHz pilot carrier omitted.

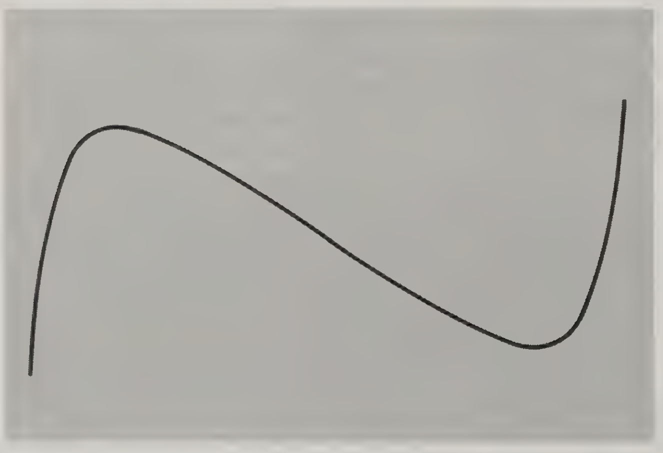


Fig. 3 Main-carrier contribution to composite signal of Figs. 1 or 2 consists of an audio sine wave at 400 Hz.

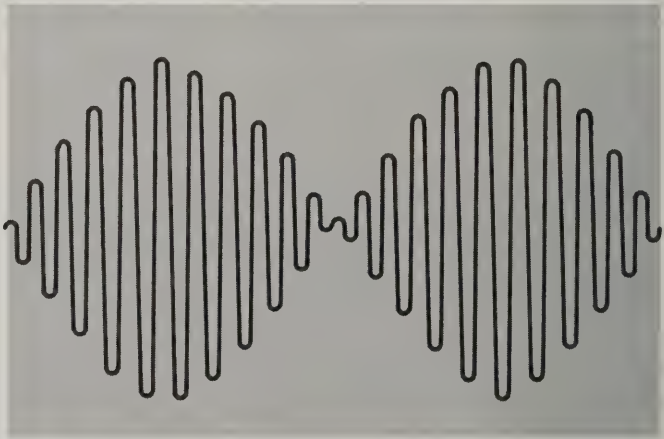


Fig. 4 Subcarrier contribution to composite signal of Figs. 1 and 2 consists of upper and lower sidebands; 38-KHz subcarrier is suppressed. Actual frequencies present are 37.6 and 38.4 KHz.

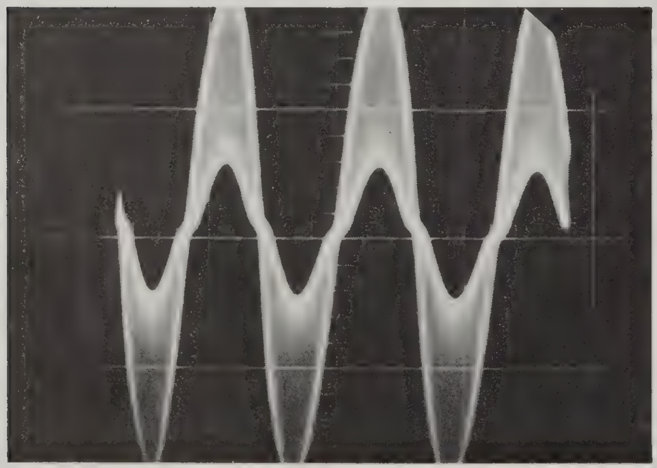


Fig. 5 Insufficient subcarrier contribution to composite signal, because of either scope roll-off or incorrect generator adjustment, results in distorted waveform shown here.

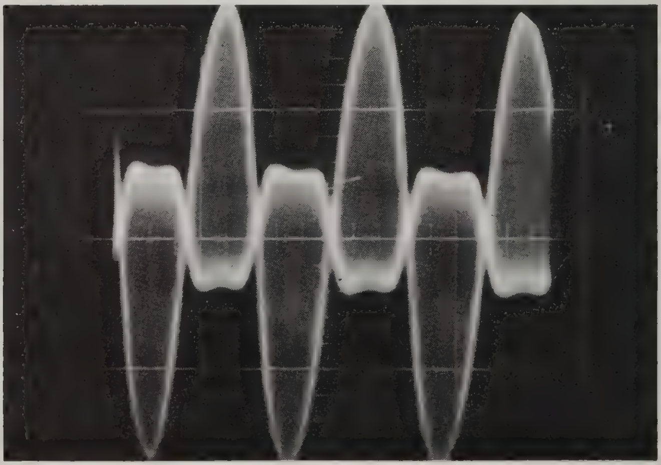


Fig. 6 Too much subcarrier contribution, compared to main channel signal, results in distorted composite signal shown here.

simple oscilloscope measurements.

Fig. 1 is a photo of a composite left-only stereo signal scoped at the output of a stereo generator. The audio frequency is 400 Hz, and riding above it is the ever-present 19-KHz pilot carrier signal. Fig. 2 shows the same display, with the 19-KHz pilot carrier turned off. When setting the level of 19-KHz signal, remember that it should be about 8 to 10 percent of the amplitude of the total composite signal, with full modulation.

The composite waveform in Fig. 2 can be thought of as consisting of a main channel sinusoidal wave, as shown in Fig. 3, plus an AM-modulated 38-KHz carrier, with the carrier itself suppressed and only the upper and lower modulation sidebands remaining, as shown in Fig. 4. When these two components are added together, the result is the waveform in Fig. 2, in which the sidebands contain frequencies above and below the 38-KHz suppressed subcarrier, depending upon the modulating audio frequency. For example, if a 15-KHz tone is to be transmitted (the upper limit of FM fidelity), the sidebands generated will be 38 KHz ±15 KHz, or 23 KHz and 53 KHz.

The scope used to observe these and other waveforms associated with stereo FM must have perfectly flat response and phase shift characteristics to at least 53 KHz, if meaningful observations are to be made. If the scope used for viewing the modulated signal previously described rolls off above, say, 20 KHz, the sideband amplitudes will be too low compared to the audio sinewave of the main channel, resulting in a waveform like that in Fig. 5. If such a waveform is seen on a scope that is known to have acceptable flat response, the trouble is in the generator itself, and corrective steps should be taken to restore a flat "baseline" to the composite signal waveform. (Methods for aligning your particular stereo generator usually will be detailed in the instruction manual supplied with the equipment, since each type of generator follows a different design scheme.)

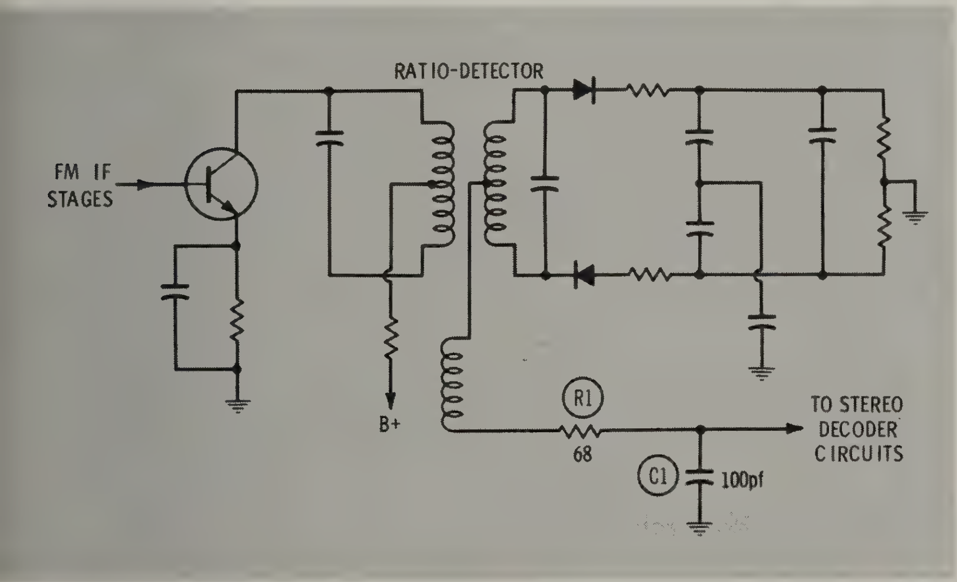


Fig. 7 Partial schematic of typical ratio-detector circuit. R1 and C1, as well as stray capacitance, can cause roll-off of subcarrier sideband amplitude.

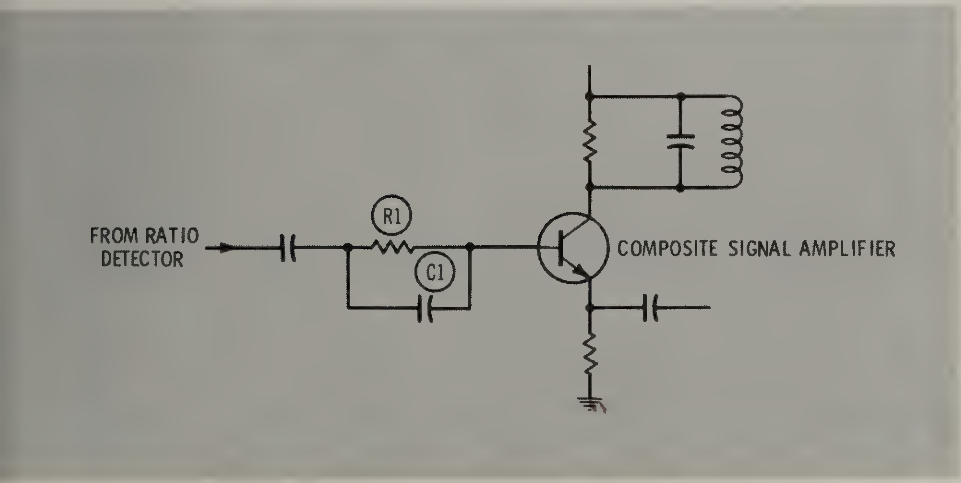


Fig. 8 Compensating network R1-C1 is often placed ahead of decoder circuit input to boost high frequencies previously attenuated at ratio-detector output.

The converse defect—low frequency response attenuation with respect to high frequency response—can cause an equally distorted composite waveform, such as that shown in Fig. 6. In this case, the "main channel" contribution is too great in amplitude with respect to the sub-channel sidebands, and the stereo FM generator should be realigned properly.

Using the Stereo Generator

The reason we don't recommend separate analysis of the decoder circuitry of a tuner or receiver, but instead suggest **overall** RF application of the stereo signal, will be apparent from the partial schematic shown in Fig. 7. This diagram shows the output circuit of the ratio detector, where the composite signal normally is obtained and from where it is fed to the decoder circuitry. The combination of R1 and C1, at the output of the ratio detector transformer, plus any stray wiring capacitances in the surrounding circuitry, can cause some attenuation of the higher frequencies contained in the recovered composite stereo signal, resulting in a detected composite signal very similar to that shown in Fig. 5. Normally, a designer faced with this problem will compensate for this roll-off in response by including an equalizing, or high-frequency emphasizing, circuit right at the input of the decoder, as shown in Fig. 8 (in which the compensating components are R1 and C2).

Other frequency and phase corrective networks also might be placed further on in the decoder circuitry. If a generator that produces a perfect composite stereo signal was connected at the input of the decoder circuit, the resulting separation measured at the left and right output terminals of the decoder would be **far from perfect,** because the decoder was actually designed to compensate for an **imperfect** composite signal, such as it normally receives from the "imperfect" IF and detector circuit preceding it. For this reason, it is much better to use the composite signal of your generator as a modulating signal for your FM generator, and then feed the modulated RF signal to the antenna terminals of the tuner or receiver under test. A complete test set-up is shown in Fig. 9.

Setting Up Proper Stereo Signal Modulation

We have already discussed the technique of setting the 19-KHz pilot carrier to the proper level with respect to audio level of the composite signal. At this point you might think that it now only is necessary to hook up the composite signal to the external modulation terminals of the FM generator and "crank up" the total composite amplitude until the modulation meter reads 75. This is not so.

Because of the complex nature of the composite signal, a conventional AC meter will not read the usual rms value (or 0.707 of one-half the p-p amplitude), but instead will read a much lower value. Thus, if you were to increase total signal amplitude until the modulation meter reads 75 KHz you would really be overmodulating the RF generator by a wide margin. The correct way to establish proper amplitudes of modulation using an external stereo composite signal involves the following step-by-step procedure:

1. Apply only 19-KHz signal, increasing the level until the RF generator's modulation meter indicates approximately 7 KHz.
2. Turn off the 19-KHz signal and apply an L-R (mono) audio signal, increasing its level until 68 KHz of modulation is indicated.
3. Turn on 19-KHz signal again and switch stereo generator mode to "L only" or "R only", without changing any audio levels.
4. In all probability, the modulation meter will now read about 45 or 50 KHz, but don't be concerned—you're actually applying full modulation of the **composite** stereo signal.

Step-by-Step Performance Analysis

Once you've established correct modulation of the FM generator with the stereo generator, the FM generator is connected to the antenna terminals of the tuner or receiver under test, and all subsequent tests and observations therefore will take into account the entire tuner, including all the complex relationships between regular tuner performance and stereo performance, and not just stereo decoder circuitry. For example, many modern stereo tuners are equipped with a threshold adjustment, which determines when the tuner will switch from "mono" to stereo reception, based upon signal strength, quieting or other factors. You can easily check this point by reducing the output of the FM generator (do **not** touch the stereo generator controls) until the stereo light goes out, or until you hear an audible indication that stereo separation has been eliminated. If too much (or little) output is required to "trigger" the switching action, it might be a fault of the threshold adjustment (if one is provided in the decoder circuitry) or it might be caused by poor sensitivity of the tuner circuitry ahead of the decoder. Further investigation will be needed to determine which is the case, using signal levels and conditions recommended by the manufacturer of the product. If you had tried to test the switching action by applying a composite stereo signal **directly to the decoder,** the poorly aligned or insensitive condition of the front end would not have been discovered.

It is a good idea to look at the detected composite waveform at the output of the FM detector in the

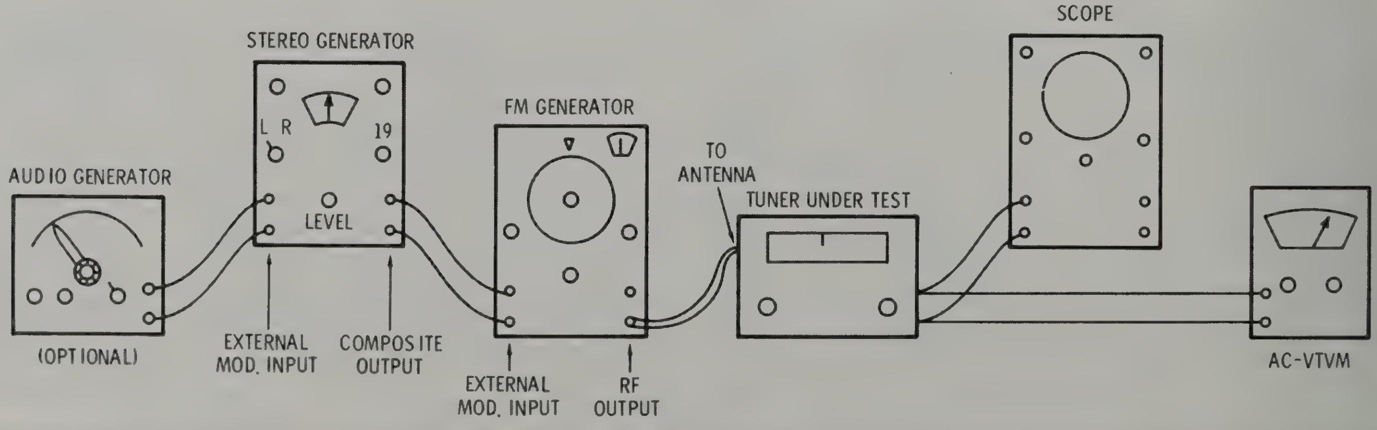


Fig. 9 Block diagram shows test setup recommended for checking and aligning the stereo portion of a stereo FM tuner or receiver.

main section of the tuner. It probably will resemble Fig. 5 instead of Fig. 6, for reasons already mentioned. After compensation (usually at the output of the first amplifying or isolating stage of the typical decoder), the composite signal probably will look more like Fig. 6 because **over-compensation** often is applied to compensate for subsequent attenuation of high frequencies in the final stereo demodulator circuitry that follows.

Other points along the circuit you should observe and adjust are the output of the 19-KHz amplifier (peaked to maximum, using a scope or, depending upon the circuitry, the stereo indicator light itself), the 38-KHz doubler tuned circuit (also peaked for maximum indication) and the 38-KHz demodulation transformer or autotransformer (peaked for maximum or for best separation).

SCA Interference

Most stereo decoder circuits are equipped with one or more filters to eliminate cross-talk from any background-music (SCA) programming, which is transmitted with the public stereo broadcast, but on a separate subcarrier, usually 67 KHz. The traps included in most stereo decoder circuitry are designed to filter out this interfering signal.

While many stereo generators are equipped to produce a 67-KHz signal, others require the separate introduction of this additional signal by means of an audio oscillator connected externally. If such is the case with your generator, the 67-KHz signal applied to it should be just enough to cause a main-carrier modulation of 10 percent (7.5 KHz), with all stereo signals temporarily removed. After this level has been established, the 19-KHz pilot carrier should be added, and the waveform present at the **output** of any SCA filters in the tuner or receiver should be observed on a scope. Adjust all SCA traps for minimum 67 KHz signal, making sure you do not mistake the 19-KHz signal for 67-KHz, since both frequencies will be present. Ordinarily, if the residual 67-KHz signal is 34 or 40 dB lower than the peak composite stereo signal, it will not be too offensive to the listener.

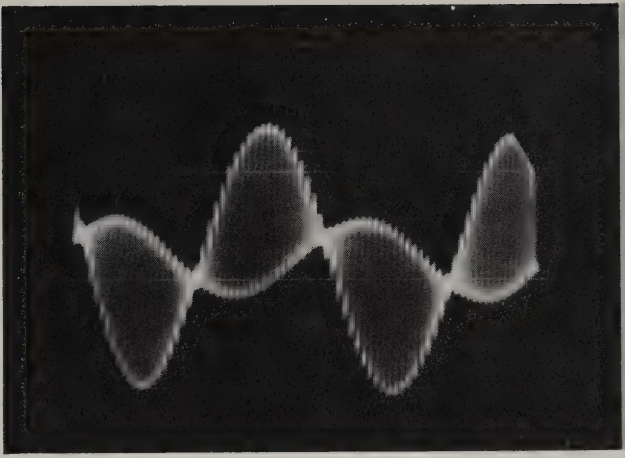


Fig. 10 Scope photo shows effect of phase error on composite signal, which often occurs at higher audio modulating frequencies. 19-KHz signal component has been omitted for clarity.

(Contrary to our earlier statement, the SCA is one adjustment that is more effectively made from "off-the-air" programs than with test equipment, because the actual SCA subcarrier is FM modulated by the secondary background music program, so that the frequency of the subcarrier is actually moving back and forth at an audio rate from about 63 KHz to about 71 KHz. The modulation of the subcarrier is what causes the "swooshing" sound which we have come to identify as SCA interference—not the mere presence of a subcarrier itself. It therefore is good practice to give the SCA traps a final touch-up while listening to a stereo station that also transmits a background music program.)

Final Separation Measurements and Adjustments

After adjusting all the tuned circuits in a stereo decoder, you might find that there also is a small trimming potentiometer which further optimizes separation. This "pot" should be adjusted while observing the "R" output of the decoder circuit with an L-only signal at some mid-frequency (around 1 KHz) applied to the system. Then reverse the procedure, observing "L" output while an R-only signal is applied. If measured separation is not the same for both channels, it is better to "compromise" the adjustments so that separation is balanced. For example, it is better to have 30 dB separation at both outputs than to have 40 dB from "L" with only 20 dB from "R".

Separation is measured by first applying an "L" signal and reading "L" output as a reference, then switching to an applied "R" signal and recording the number of dB below the first reading that is now obtained. You can repeat separation measurements at all audio frequencies, from 50 Hz to 15,000 Hz, but don't be surprised if the separation readings obtained at the frequency extremes are much poorer than those obtained at mid-band. This is particularly true at the higher frequencies where, in addition to amplitude discrepancies between the main-channel and sub-channel components, there also might be **phase** discrepancies.

An example of phase error between main and sub-channel components of the stereo composite signal is shown in Fig. 10. Such a waveform also will cause poor separation. In fact, the FCC, in specifying tolerances for stereo FM broadcasting, insists that the phase error between main- and sub-channel components not exceed 2 degrees for any audio frequency from 50 Hz to 15,000 Hz. Maintaining such limits, as well as an amplitude discrepancy no greater than 3.5 percent, assures separation of at least 30 dB at all frequencies used.

Summary

The right stereo generator, used in conjunction with a good FM signal generator, can reduce stereo multiplex servicing to a few simple waveform observations and adjustments.

Troubles in stereo FM receivers often are the result of faults in the tuner section rather than in the stereo decoder circuitry. The ability to analyze tuner performance along with stereo performance as they relate to each other is essential for effective servicing of modern stereo high-fidelity systems. ▲

Chapter 8

Cartridge and turntable measurements and tests

Cartridge and turntable measurement tests are required to obtain and verify performance within accepted high-fidelity limits. These tests are listed below.

Cartridge Tests

1. Measurement of needle pressure
2. Checking of cartridge output voltage
3. Hum Tests
4. Test of frequency response
5. Measurement of tracking distortion
6. Verification of stylus angle
7. Observation of stylus tip
8. Verification of tracking angle

Turntable Tests

1. Measurement of turntable speed
2. Check for wow
3. Check for flutter
4. Level test
5. Tests for wobble and/or eccentricity
6. Approximate measurement of rumble.

Measurement of Needle Pressure

Fig. 1 shows a typical needle-pressure gauge. Other types may be used, but all indicate the pressure in grams.

Normal needle pressures are different for various types of recordings. A typical hi-fi, super lightweight, pickup employs 1, 2.5 or 3 mil radius sapphire tips. (One mil equals 0.001 inch.) A 25-ohm pickup of this type uses a nominal needle pressure of 9 grams, ±1 gram. On the other hand, a 400-ohm pickup operates with a nominal needle pressure of 5 grams, ±1 gram. For precise evaluations, stylus pressure must be checked within specified tolerance limits against tracking distortion. This measurement is discussed later in this article.

The needle-pressure gauge should be placed so that the indication is obtained at the normal playing level of the pickup. This level varies in most changers; therefore, measurement is made at the average playing level. Although the pressure usually varies as the pickup is raised, and decreases as the pickup is lowered, this variation should not exceed 25%. If a greater variation is measured, look for a defect in the arm assembly. Obscure defects must be corrected by replacement of the arm assembly.

Checking of Cartridge Output Voltage

Cartridge output voltage is checked with an AC VTVM or with a calibrated oscilloscope. The advantage of a scope is that hum voltage can be distinguished from the normal cartridge output. Only the VTVM method will be discussed in this section.

The test setup for checking cartridge output voltage is shown in Fig. 2. It is desirable to employ a test record that provides a constant level at a standard test frequency, such as 1 KHz or 400 Hz.

A typical hi-fi super lightweight pickup that works into a 25-ohm load (R in Fig. 2) normally provides about 10 mv rms. If the output is weak, it is advisable to check the stylus tip, as explained later. If the tip is in good condition, but output is weak, the cartridge should be replaced.

The 25-ohm pickup described previously is often used with a 1-to-5 ratio transformer. Normal loading of the transformer results in a secondary output voltage of approximately 300 mv rms. On the other hand, a 400-ohm pickup normally is used with a 1-to-6 ratio transformer. In this case, about 40 mv rms output will be measured across the secondary. Note that the secondary must be connected to the correct value of load impedance to obtain a meaningful voltage measurement.

With reference to Fig. 3, Np denotes the number of primary turns, and Ns denotes the number of secondary turns. The winding ratio is equal to Np/Ns.

The voltage ratio of the transformer is the same as the turns ratio (except for losses due to leakage flux under normal load). The current ratio of the transformer is the inverse of the turns ratio (except for losses). The impedance ratio is equal to the square of the turns ratio. This means that if the cartridge has an impedance of 400 ohms, and the transformer winding ratio is 1-to-6, the impedance ratio is 1-to-36, and the secondary impedance is 14,400 ohms. Thus, the nominal secondary load resistor has a value of 14,400 ohms. Note, however, that a somewhat different value of secondary load impedance might be chosen to compensate for incidental transformer characteristics such as distributed capacitance. This topic is explained in greater detail in this article under "Test of Frequency Response."

The normal output voltages of common types of cartridges may be summarized as follows: A variable-reluctance cartridge will provide from 10 to 30 mv rms; a ceramic cartridge, 500 to 1200 mv rms; and crystal cartridges, 0.5 to 5 volts rms. These values assume that the cartridge works into its specified load.

The load is an impedance which consists of both the load resistance, cable capacitance, and any leakage reactance of the transformer primary. The variable-reluctance pickup works into a load impedance of

5,000 to 50,000 ohms, with a stylus pressure of 16 grams or slightly more. The ceramic pickup works into a load impedance of approximately 1 megohm, with a stylus force of about 22 grams. The crystal pickup works into a load impedance as high as 5 megohms, with a stylus force of less than 70 grams.

Hum Tests

Pickups that employ iron and coil construction are more likely to be affected by stray hum fields than crystal pickups. Of course, a poorly shielded lead from a high-impedance pickup is susceptible to stray electrostatic hum fields. Pickups with iron and coil construction are generally used with solid-steel turntables, because magnetic shielding is thereby provided against stray hum fields from the motor and power supply.

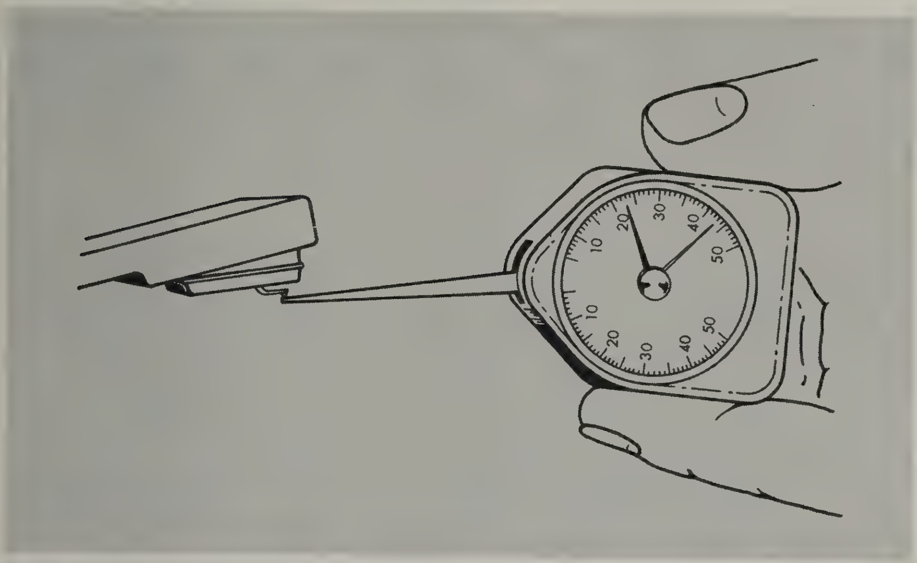


Fig. 1 A typical needle-pressure gauge used for measuring the pressure exerted at the tip of the needle.

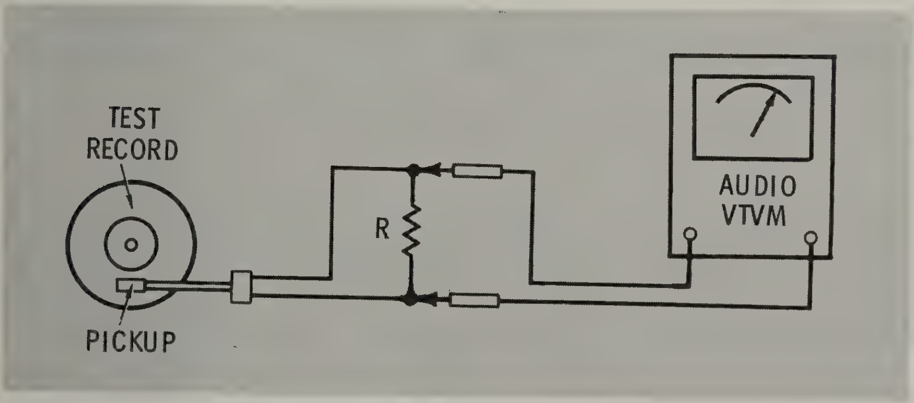
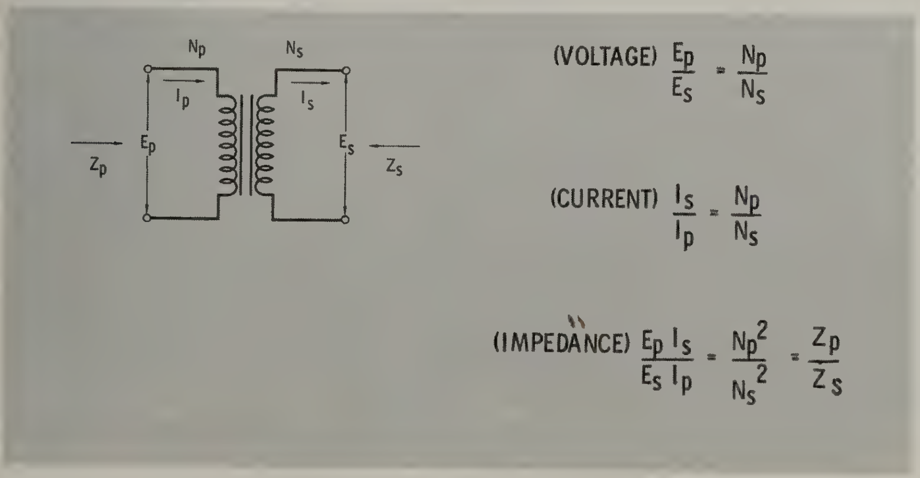


Fig. 2 Test setup for checking cartridge output voltage.



To check for hum, use the test setup shown in Fig. 2, with the pickup held slightly above the record surface. Any reading on the meter is probably the product of hum pickup.

A better test for hum is made by using a scope in place of the VTVM in Fig. 2. Hum voltage is then seen as a 60-Hz (or 120-Hz) waveform, and can be definitely identified. Note that 120-Hz hum is most likely to stem from stray fields of a filter inductor. When playing a test record that has a 1-KHz tone, hum shows up with the tone signal superimposed on the low-frequency waveform, as shown in Fig. 4. The hum level can be minimized in any case by correction of shielding defects to prevent stray fields from gaining access to the pickup.

Test of Frequency Response

Frequency response is checked with a test record and an audio VTVM, using the setup shown in Fig. 2. Various types of test records are available. A **gliding-tone** test record can be compared to an audio sweep generator. One standard test record starts at 14 KHz and ends at 10 Hz. Another starts at 10 KHz and ends at 50 Hz. A **banded-tone** test record provides only a 1-KHz tone in steps with increasing output levels. This type of record is suitable for harmonic-distortion tests, but not for frequency-response checks. Some test records provide both banded-tone and gliding-tone outputs. For example, a banded-tone sequence may be followed by alternate gliding and constant audio frequencies from 3 KHz to 30 Hz. **Two-tone** test records are used only for IM distortion measurements.

The frequency response of a good ceramic or crystal cartridge should be reasonably uniform from 50 Hz to approximately 15 KHz. Variable-relutance cartridges should have reasonably flat frequency response to approximately 20 KHz. Some types of high-output cartridges have an upper frequency limit of 8 KHz. Although one or more peaks in frequency response will often be found, the peak amplitude(s) should be less than 2 dB. An audio VTVM with a dB scale is particularly useful for these tests.

Load-impedance values can have considerable effect on the frequency response of a ceramic or crystal cartridge. Poor frequency response can sometimes be improved by a change in load resistance and/or capacitance. Equalizer circuits for standard recordings are shown in Fig. 5. If the high-frequency response of a pickup drops off gradually, this characteristic will often be acceptable, inasmuch as the pre-emphasis employed in most recordings must be equalized in the hi-fi system.

Measurement of Tracking Distortion

An ideal tone arm and pickup would track a record (see Fig. 6A) so that the stylus would always move perpendicularly to the tangent at the record groove. In practice, this can occur only at one groove in the record because the pickup moves on a curve from the outer to the inner grooves, as shown in Fig. 6A. A reasonably good test of the tracking angle is to place the stylus at the outer grooves, then at an intermediate groove, and finally at an inner groove, and to observe whether the edge of the cartridge is **reasonably** tangent to the grooves.

A tracking error of 3° or 4° is normally expected at the outer and inner grooves. Large tracking errors are plainly visible, but small errors must be detected by careful measurement. Excessive errors in the tracking angle produce second-harmonic distortion, as shown in Fig. 6B. Harmonic distortion is measured with the test setup shown in Fig. 7. In such tests, it is assumed the preamplifier has been checked previously for distortion, using a quality audio oscillator.

Note that the percentage of harmonic distortion measured in Fig. 6B is not entirely accounted for by tracking error and possible preamplifier distortion. That is, the pickup might have inherent harmonic distortion. At this point in the procedure, we are concerned with minimizing the harmonic distortion due to tracking error. Since this is a function of stylus pressure, it is advisable to try the effect of increasing the stylus pressure within reasonable limits. Some pickups are rated for minimum stylus force to provide optimum tracking. If a recording or turntable is warped, or slightly eccentric, a 50% increase in stylus force may be required to obtain

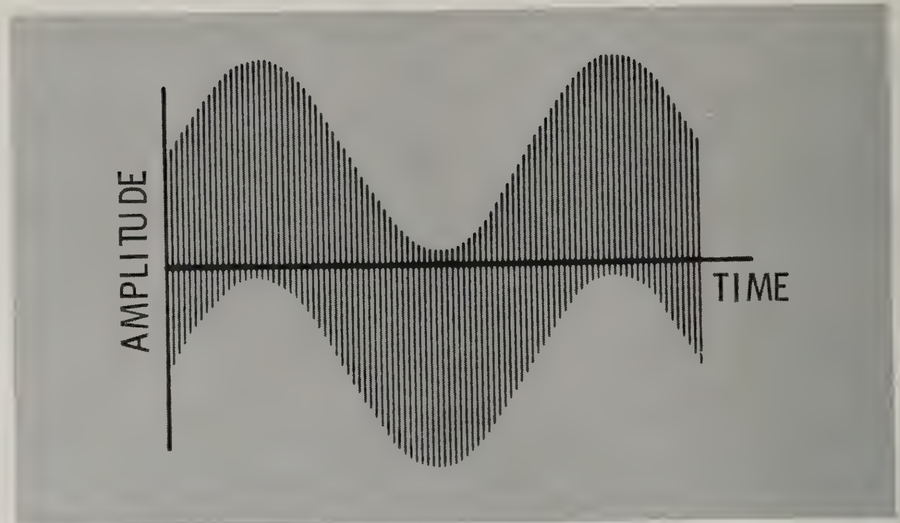
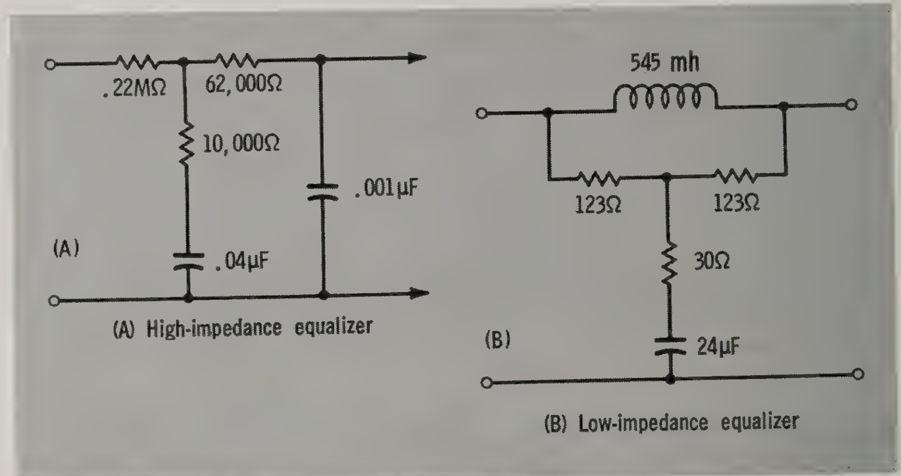


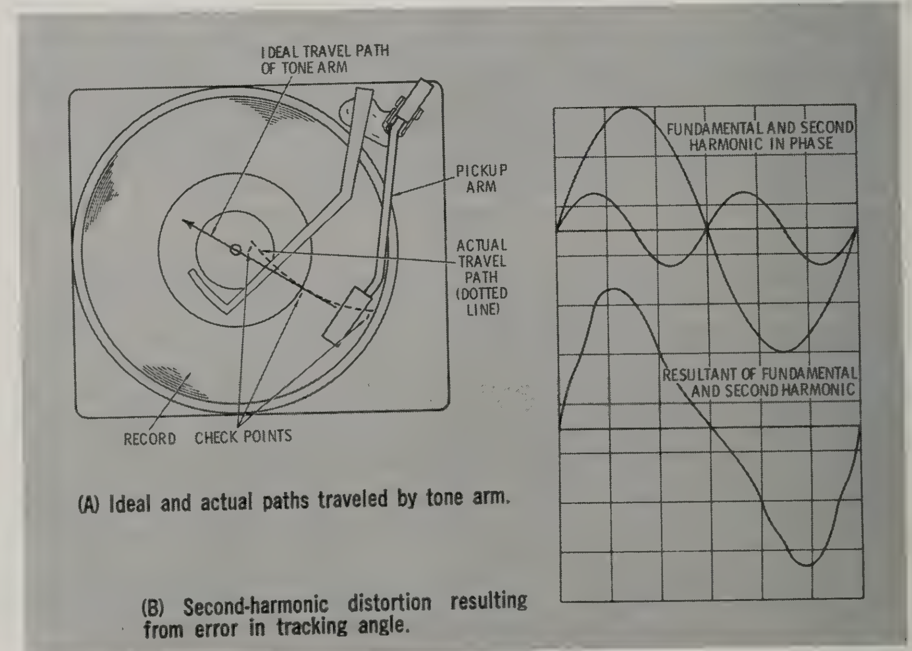
Fig. 4 Hum voltage superimposed on low-frequency waveform.



(A) High-impedance equalizer

(B) Low-impedance equalizer

Fig. 5 Equalizer circuits can be used to improve frequency response.


IDEAL TRAVEL PATH OF TONE ARM
PICKUP ARM
ACTUAL TRAVEL PATH (DOTTED LINE)
RECORD
CHECK POINTS
FUNDAMENTAL AND SECOND HARMONIC IN PHASE
RESULTANT OF FUNDAMENTAL AND SECOND HARMONIC

(A) Ideal and actual paths traveled by tone arm.

(B) Second-harmonic distortion resulting from error in tracking angle.

Fig 6 Illustrations showing tracking characteristics and harmonic distortion resulting from improper tracking.

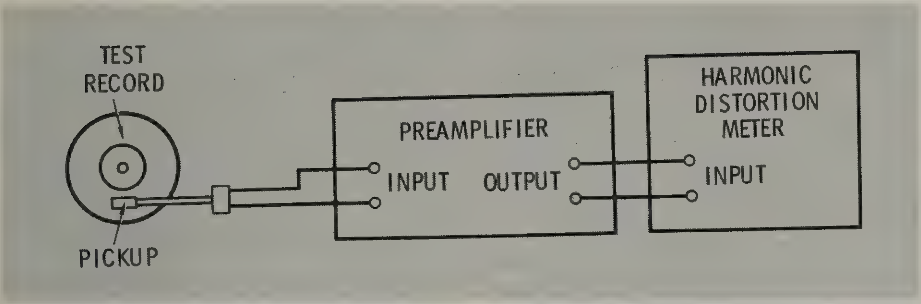


Fig. 7 Test setup for measuring harmonic distortion.

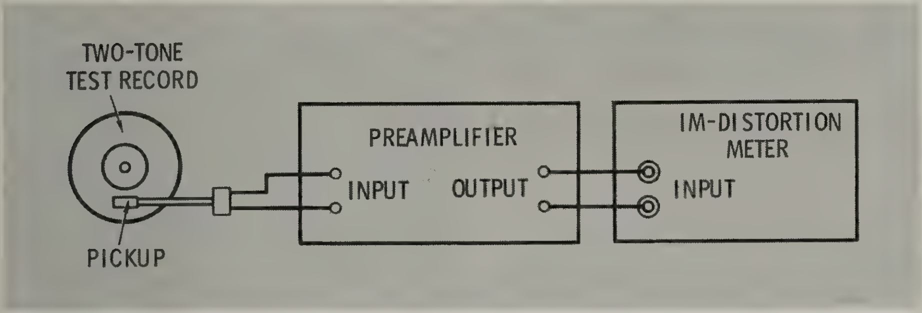


Fig. 8 Test setup for measuring intermodulation distortion using two-tone test record.

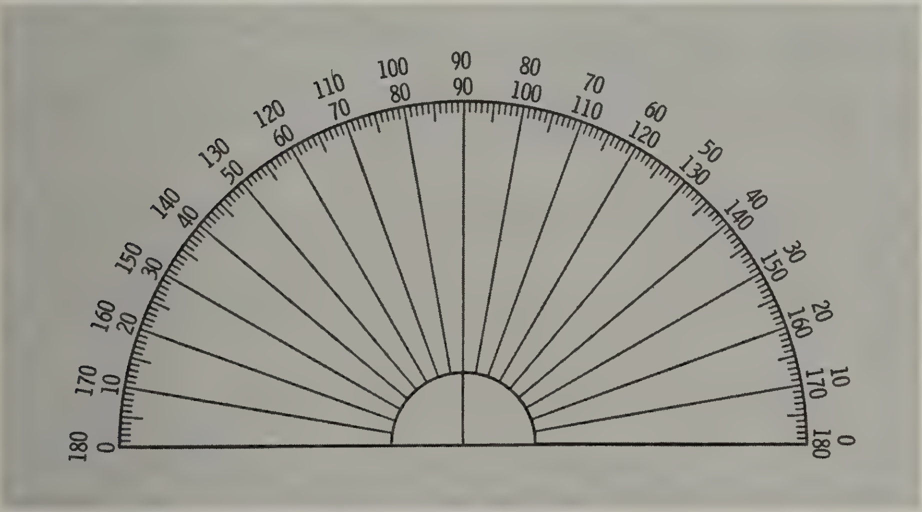


Fig. 9 Protractor can be used for measuring stylus angle.

Fig. 10 Fifty-power microscope for visually inspecting condition of stylus.

good tracking.

After checking for harmonic distortion, it is good practice to make an intermodulation distortion test, as depicted in Fig. 8. The two-tone test record must provide suitable frequencies. Assuming that the preamplifier has been checked previously and found to have acceptably low IM distortion, an objectionably high meter reading is due to nonlinearity in the pickup. A banded-tone 1-KHz test record with progressive output levels may show that IM distortion is negligible except at high output level. In any case, a defective pickup must be replaced.

Verification of Stylus Angle

Even if the needle pressure is optimum, distortion will occur if the stylus angle is incorrect. Therefore, it is good practice to check the stylus angle against the manufacturer's specifications, if available. Angles can be conveniently measured with a protractor, such as depicted in Fig. 9. If the stylus angle is out of tolerance because of mechanical distortion, it is generally impractical to correct the trouble—replacement should be made at the outset.

Observation of Stylus Tip

Unless a new stylus is used in making performance measurements, it is a good practice to inspect the stylus for wear with a 50-power stylus inspection microscope, as shown in Fig. 10. If the stylus shows signs of wear, it should be replaced before playing a test record. By the same token, a test record should be checked periodically for harmonic and IM distortion, using a good stylus. When the test record begins to show appreciable distortion, it should be discarded.

Measurement of Turntable Speed

A turntable must rotate at constant speed, even during lightly and heavily recorded passages. Turntable speed is conveniently measured with a stroboscopic disc illustrated in Fig. 11. The strobe disc is viewed under a neon or fluorescent light powered from a 60-Hz source. One of the barred circles will appear to stand still, or to rotate slowly clockwise or counterclockwise. Motionless bars indicate the calibrated

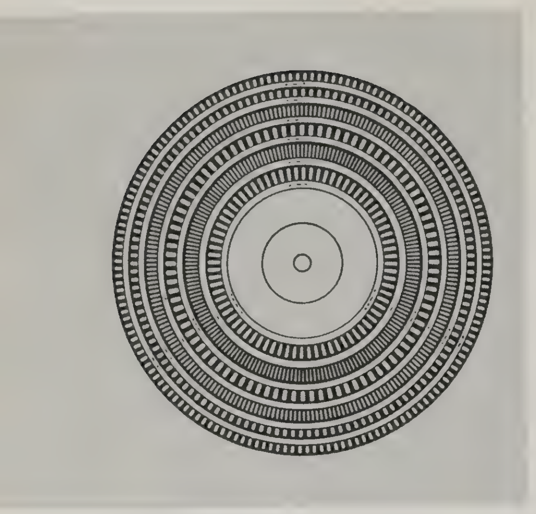


Fig. 11 Stroboscopic disc for measuring turntable speed.

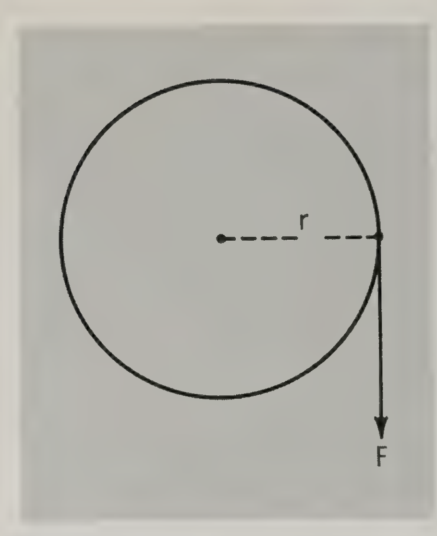


Fig. 12 Illustration of twisting force called torque.

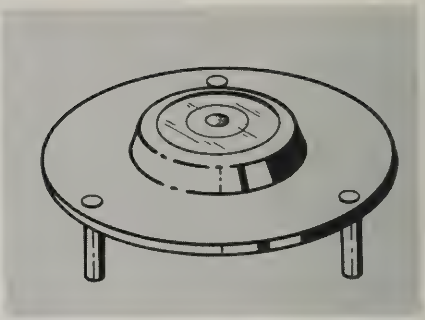


Fig. 13 Device for leveling a turntable.

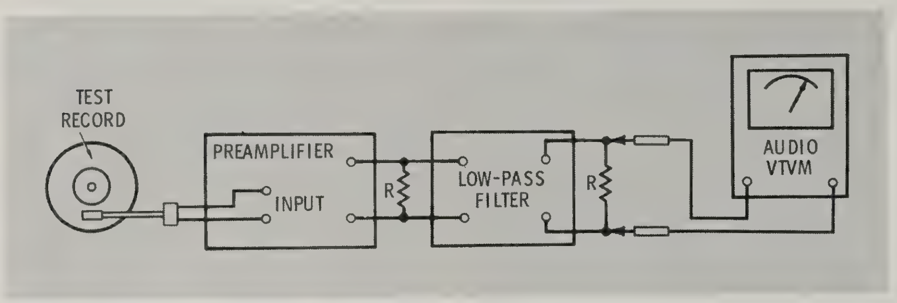


Fig. 14 Test setup for measuring rumble.

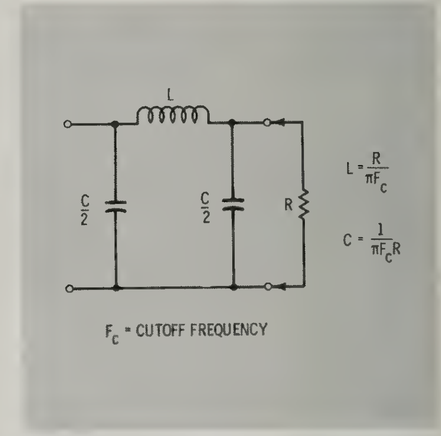


Fig. 15 A single-section, low-pass Pi filter used in rumble test.

speed of the particular barred circle. Clockwise rotation indicates that the actual speed is slower.

In some cases, the barred circle will appear to stand still momentarily, rotate for a while, and then to stand still again. This trouble is sometimes caused by oil or other foreign substances on the drive wheel. These parts can be cleaned with alcohol.

The same trouble symptom can be caused by defective turntable or motor bearings. Also, there may be unsuspected mechanical interference, such as the turntable rubbing against some surface or object.

Wow

If a turntable does not rotate at constant speed, the condition is technically termed **wow.** It is basically a mechanical form of frequency modulation. Economy-type turntables may be unsatisfactory for hi-fi reproduction because of marginal motor power and lack of sufficient flywheel effect. A strobe disc test under playing conditions will disclose wow which, in this case, cannot be eliminated.

Motor and turntable units are often specified by manufacturers for motor type, power consumption, and the torque required to brake the turntable from a given speed to a lower speed (such as from 78 rpm to 77 rpm). Torque denotes a twisting force, as depicted in Fig. 12. The magnitude of torque is expressed as the product of force and distance (F and r in Fig. 12.) Braking torque for a turntable is stated in ounce-inch units.

Of course, wow will become evident if the motor is operated at subnormal line voltage. Noticeable wow will be observed on heavily recorded passages. Another obvious cause of wow is a warped record; the tolerable limit of warp is about 1/16 inch. Special instruments are available to measure wow, and indicate the percentage in rms deviation of a tone frequency with respect to its average frequency. However, a strobe disc serves the same purpose for routine shop tests. The only requirement in strobe-disc application is that the changes in apparent speed of bar rotation be observed carefully. When wow is slight, it might be overlooked in an off-hand test.

Flutter

Flutter is related to wow, but has a comparatively higher deviation rate. The human ear is more sensitive to flutter than to wow. Flutter is usually caused by small defects in the motor or the mechanical drive system. It tends to occur in units that have been in extended service, with little or no attention to lubrication. The resulting wear causes poor mechanical fit, which must be corrected by replacement of the faulty parts. Although flutter meters are available, a strobe disc serves adequately for routine shop tests. Both wow and flutter meters are comparatively expensive instruments, and their cost can be justified only by the larger shops or labs.

Understandably, a turntable must be level to provide optimum performance. A typical turntable level is illustrated in Fig. 13. The turntable must also be free from wobble, and should maintain its level against moderate vertical pressure near the edges. If a wow and/or flutter meter is available, note that a meter reading of less than 1% is considered acceptable. A good hi-fi turntable may show as little as 0.1% wow or flutter, even on heavily recorded passages.

Rumble

Both wow and flutter are distinguished from **rumble.** Rumble consists of a series of random pulses (not a cyclic low-frequency interference). It is caused by vibrations within the player, and is often decribed as noise "like furniture moving around upstairs." A scope test quickly shows whether interference has a rumble waveform—the pattern is random and does not have a steady cyclic form in the case of rumble. Although rumble can be reduced or eliminated by means of an amplifier rumble filter which cuts off sharply below 50 Hz, it is obviously desirable to correct the cause of the rumble if it is practical to do so.

When rumble is a problem, check the motor for vibration. Any mechanical defect that causes vibrations to be transmitted to the turntable can cause rumble. Note that rumble can be measured, using the test setup shown in Fig. 14. The low-pass filter has a cutoff frequency of approximately 300 Hz. The test procedure is as follows:

First, the low-pass filter is removed from the circuit, and the output is measured using a 1-KHz test record. The preamplifier controls are adjusted for maximum-rated output and flat frequency response. Then, the low-pass filter is connected into the test circuit, and the VTVM sensitivity is advanced to obtain a convenient reading. Finally, lift the pickup from the test record and note any change in VTVM reading. The amount of rumble is expressed as the number of dB below maximum rated output from the preamplifier.

Let us briefly consider the component values used in a low-pass filter. With reference to Fig. 15, choose a cutoff frequency, fc, and a terminating resistance, R. Then find the required values of L and C as indicated. For example, if fc=300 Hz, and R=75 ohms, then L=80 mh and C=0.071 mfd. One-half of C, or 0.035 mfd is connected at each end of L to form the filter. A sharper cutoff characteristic can be obtained by connecting several filter sections in series. (Note that the inductors of the multi-section filter must not couple into one another.)

Conclusion

Although other types of tests are made in audio laboratories, these are generally out of the question for the service shop because of the costly test equipment that is required. For example, measurement of the static compliance of a stylus requires a calibrated shadowgraph and a sensitive balance. However, the simpler tests that have been explained in this series of articles permit almost any service problem to be solved with a minimum of time and effort. Apprentice technicians are often awed by ordinary harmonic and intermodulation distortion meters, and occasionally by a scope. However, practice makes perfect, and the modern shop cannot hope to remain competitive unless modern test equipment is utilized. ▲

Chapter 9

Stereo FM auto radio servicing

Major Systems in Stereo FM Radios

Because most stereo FM receivers are relatively complex pieces of equipment, it is best to divide them into their component systems before outlining related troubleshooting procedures. In this chapter, we will divide a typical system into the following categories: tuner (or front end), IF amplifier, detector, multiplex decoder and the stereo audio amplifiers.

Tuner

The tuner is the section which amplifies the various FM broadcast signals fed to it from the antenna, and then converts them to a lower, or intermittent frequency, (IF) usually 10.7 MHz in domestic FM sets. After conversion, the signal is then fed to the IF amplifier chain.

The so-called standard FM tuner consists of three transistors: an RF lator, as shown in Fig. 1. This parlator, as shown in Fig. 1. This particular tuner is used in the 1970 Ford Galaxy FM stereo car radio (Model OFBF), manufactured by Bendix.

The design of the RF amplifier in Fig. 1 follows the practice of using the common-base configuration for VHF amplifiers. Besides producing higher gain at these frequencies than some other configurations, it also requires little or no neutralization. If a common-emitter stage were used, neutralization would probably be necessary, to prevent the stage from oscillating.

The local oscillator circuit in Fig. 1 is a relatively standard circuit which uses a varactor for AFC correction.

The circuit in Fig. 2 is a newer version of the FM RF amplifier. This type of circuit, employed in both hi-fi FM tuners and at least one AM/FM car radio, uses an IGFET (insulated-gate, field-effect transistor). The IGFET's extremely high input impedance and it's cross-modulation characteristics have made it attractive to FM-tuner designers during the past several years. Mass production and technological advances have lowered IGFET prices so that they can be used even in some relatively inexpensive FM sets.

The IGFET is extremely sensitive and must be handled properly to prevent tool and body static charges shorting out the gate insulation. Most replacement IGFET's come packaged with either a metal shorting ring binding the wire leads or a piece of lead foil sheathing them.

IF amplifier

The IF amplifier strip is a tuned-RF type of amplifier. It differs from the RF amplifier in the tuner in that it is tuned to only one frequency. Except in a few sets imported from Europe, this frequency is 10.7 MHz.

There are, generally speaking, only two basic types of IF amplifiers used in modern Stereo FM equipment. The first, and older of the two, is shown in Fig. 3. It is a standard bipolar transistor circuit. In actual practice, three or four such stages make up the IF amplifier section.

The circuit in Fig. 4 is a version of the more modern type of FM IF amplifier, which uses integrated circuits (IC's). This particular circuit uses the RCA CA3012 IC. Several variations of this circuit have been used in a number of popular-brand FM stereo tuners and receivers (Further information about this and other types of IC circuits is available in a book titled "Linear Integrated Circuits", which is available from RCA distributors.)

The detector

The FM detector stage is one of those areas in which a good memory is a real asset. How many technicians can, for example, remember the fine points of difference that exist between the Ratio Detector and the Discriminator?

Another type of detector, the integrated-circuit quadrature type, is used now in hi-fi and in at least one brand of auto radio. TV men should be familiar with this system. Hi-fi and auto radio technicians, on the other hand, might have only recently encountered these circuit designs. Certain Delco stereo FM car radios have used this type of detector for the past three years.

The three basic types of detectors are shown in Figs. 5A, 5B and 5C. Several semiconductor manufacturers presently offer one or more IC's that include the limiter, two or more diodes for the detector, and, sometimes, even the last FM IF amplifier transistor. So don't assume that an IC in the last stage means that a quadrature detector is being used. The quadrature can be recognized by the presence of a phasing coil. Other types of IC FM detectors use a regular detector transformer or have no tuned circuit at all except in the input network.

Multiplex circuitry

The multiplex section processes the audio recovered by the FM detector (which contains both L+R and L–R signals), producing right and left separation.

Although a complete discussion of multiplex systems, processes, and circuits is beyond the scope of this article, I will briefly review the process.

The encoded L–R signal is a double-sideband, suppressed-carrier

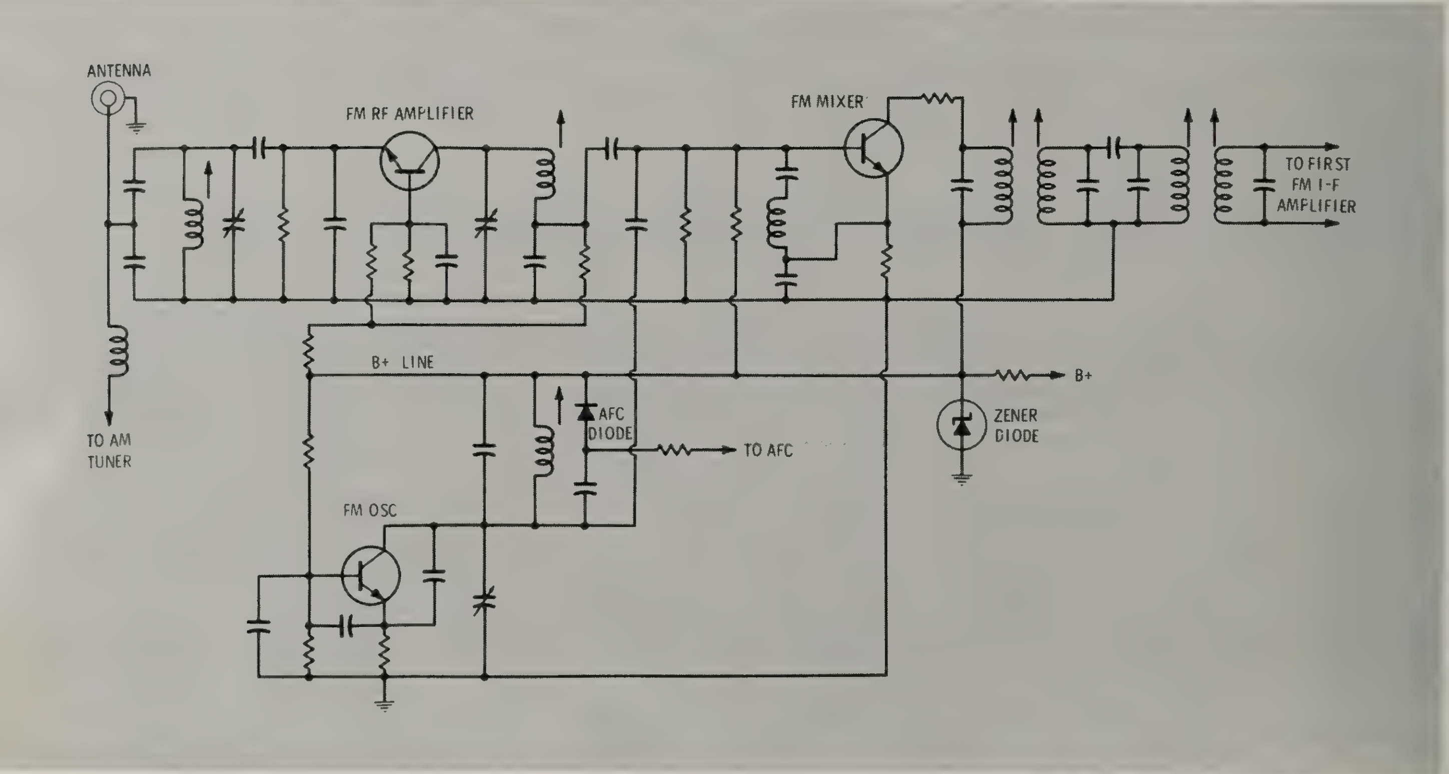


Fig. 1 FM tuner section used in Ford Model OFBF radio. Common-base configuration of FM RF amplifier stage requires little or no neutralization to prevent oscillation. Varacter provides AFC action.

(DSB) form of modulation. (DSB is a variation of AM.) The frequency of the associated carrier (called a "sub-carrier", to distinguish it from the FM main carrier) is 38 KHz. Because this subcarrier is suppressed at the transmitting station, the receiver must generate and reinsert a subcarrier. Fortunately, there also is a 19-KHz pilot signal transmitted along with the L+R and L–R signals. By one of several methods, this 19-KHz pilot signal is used to generate a 38-KHz subcarrier which, in turn, is used to decode the two stereo channels.

The whole stereo process is explained in several ELECTRONIC SERVICING articles and in a Howard W. Sams book titled "FM Multiplexing For Stereo" (Catalog No. 20199), by Leonard Feldman.

Audio

The audio stages used in stereo FM receivers require little discussion because most are of common design. Consequently, the usual techniques used to isolate defects in conventional audio circuits also apply to those used in stereo FM receivers.

Troubleshooting Procedures

Tuner check-out

After connecting the power and speaker leads and any other leads required, adjust the controls to determine if proper operation can be achieved, and, if not, what the trouble symptoms are. Recognition of symptoms such as a high level of hiss, a blinking stereo-indicator lamp, or low volume with good sensitivity can help isolate the defective stage even before the set

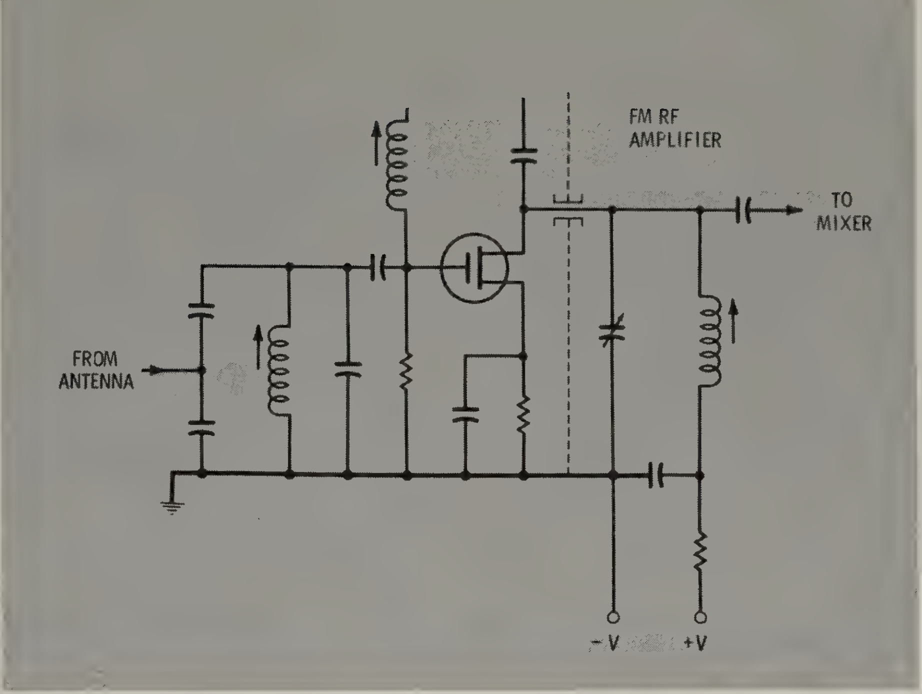


Fig. 2 Newer type of FM RF amplifier design uses insulated-gate, field-effect transistor (IGFET), which offers extremely high input impedance and is relatively immune to cross modulation. Because tool and body static can short out the gate insulation of IGFET's, most replacement units are packaged with leads shorted together.

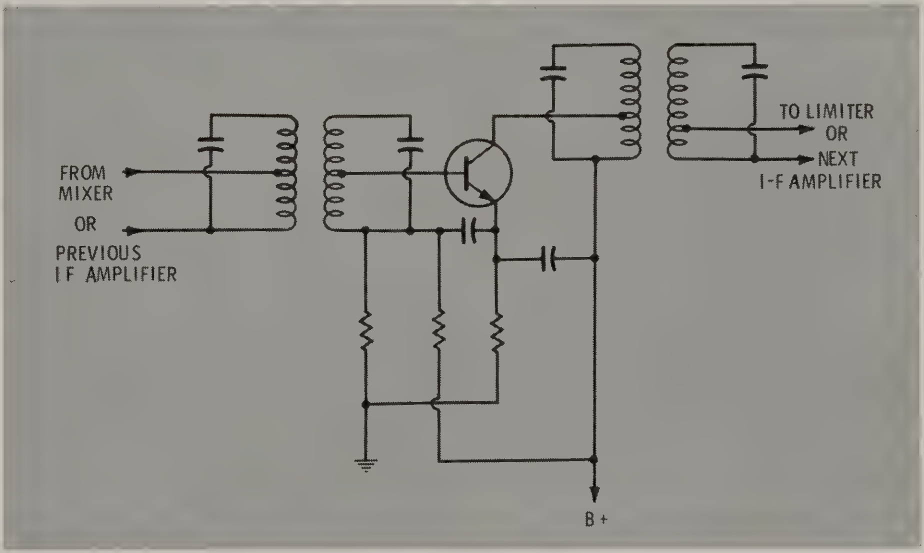


Fig. 3 Older design of FM IF amplifier, which uses conventional bipolar transistor. This basic design is still used in some stereo FM equipment. Three or four of stage shown here make up the IF amplifier section.

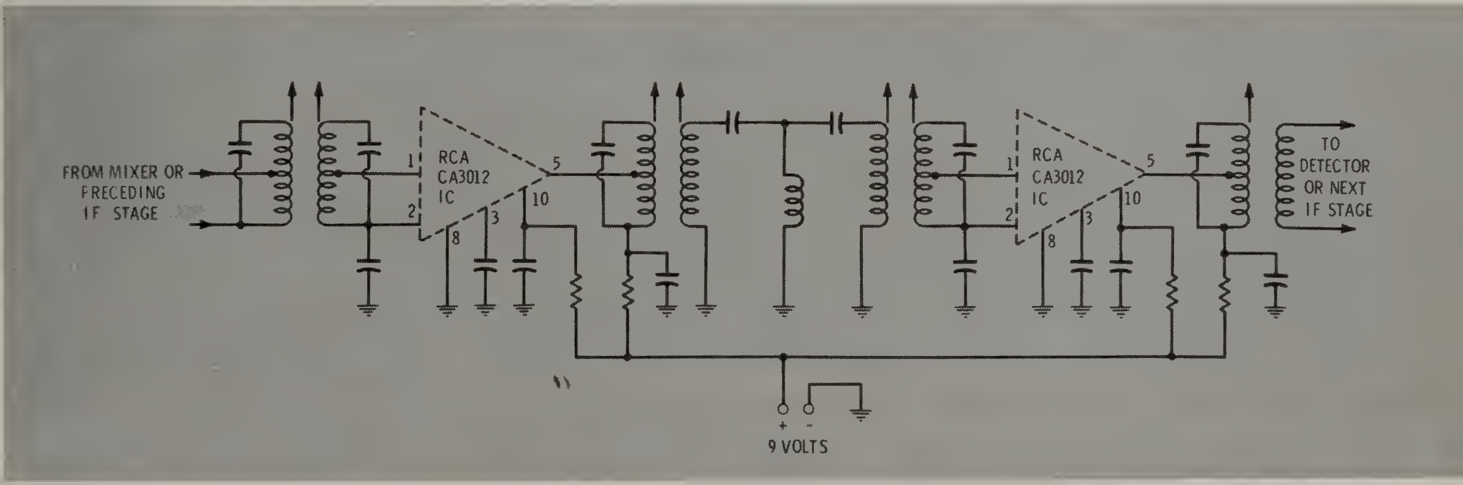


Fig. 4 Newer FM IF amplifier design uses integrated circuits (IC's).

is warm.

Because of the more sensitive nature of VHF signals, the front end of an FM set is especially difficult to service. For example, test equipment probes often present considerable stray capacitance to the tuned circuits, which completely changes their response, or range of operation, particularly that of the FM local oscillator. Because of this, many otherwise routine tests are useless in this stage. Probably the most reliable method of determining whether the oscillator is operating is to check the bias on the oscillator transistor with a high-impedance DC voltmeter. Even though the frequency of the oscillator might have shifted or the function stopped completely, the bias probably will not be affected. This isn't always true, but is often enough to make it a useful indication.

An indirect check of the FM oscillator can be made by using a grid-dip meter for signal substitution, as shown in Fig. 6. I use a low-priced, kit-type meter. Although the output of the grid-dip meter is relatively unstable at VHF frequencies, it is sufficiently stable for this purpose. Tune the grid-dip meter to the approximate frequency of the FM oscillator (radio dial frequency plus 10.7 MHz). Then loosely couple the grid-dip meter coil to the FM mixer circuit. This usually requires placing the coil physically close to the FM mixer transistor. Sometimes, this method will work a foot or more from the set. Don't, however, depend on it. I did—only **once.** I assumed an oscillator was bad when, in fact, the grid-dip meter wasn't close enough to produce a recognizable effect.

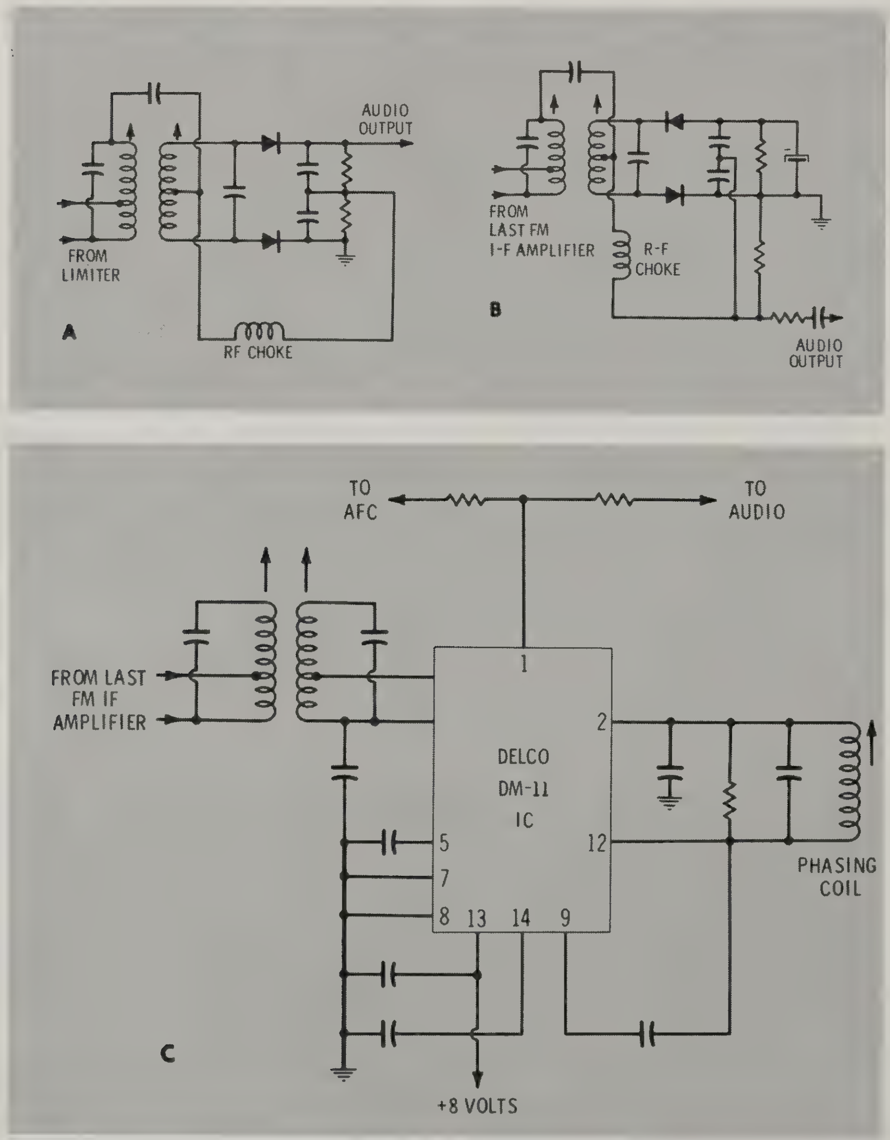


Fig. 5 Three basic types of FM detectors. A) Foster-Seeley design is usually preceded by a limiter because it is sensitive to any AM variations of FM signal. B) Ratio detector need not be preceded by a limiter stage because it has "built-in" AM suppression. C) Quadrature detector. The phasing coil, not use of IC, is best indication of quadrature design.

If a defective local oscillator is the cause of abnormal operation, the receiver will pick up stations as you tune the grid-dip meter through the range of frequencies normally covered by the local oscillator of the receiver. The grid-dip meter will cause hetrodyning and quieting if another stage in the front end of the receiver is defective.

In areas where the local stations produce a high signal level, occassionally a receiver with poor sensitivity will come to life when the shielding covers are removed. In areas of high signal levels, certain sensitivity problems can be "masked" unless the set is examined while tuned to a weaker station.

Loose mounting screws are common causes of poor sensitivity and other tuning problems. This is especially true in automotive equipment. Loose screws cause a wide and varying range of symptoms. Ground loops and open DC grounds seem to be the most common faults directly attributable to loose mounting screws. In some cases, the tuner will merely shift frequency, as if the AFC were at fault. In others, the loose mounting screws will cause a high level of static. In still others, there will be no FM reception. Although this problem does occur in some sealed tuner designs, it is more common in sets which use an open FM printed-circuit board that is bolted to the chassis.

The AFC circuit can usually be checked by tuning the receiver slightly off station with the AFC disabled. (Many sets provide a switch to disable the AFC). When the AFC is turned on again, the receiver should be automatically tuned to the exact center of the broadcast signal.

Listening to fringe-area or other very weak signals is one excellent way to evaluate the performance and condition of the FM tuner and IF strip. This type of test is even more meaningful if the technician is familiar with how such stations sound and act on a normally operating receiver of comparable design and quality. Note, for example, whether the stereo-indicator lamp lights and whether the signal is clear or fades or varies continuously. If the set uses an AC line cord or 300-ohm, twin-lead dipole as an antenna, make the preceding checks using that antenna. If the gain of the shop antenna is substantially higher than that of the receiver's antenna, it could mask a weak condition in the receiver.

The preceding tests are far too subjective to be considered absolute and infallible. They are useful, however, for making rapid judgements on which an estimate is to be based, or when trying to decide where to start searching for the defect.

Subjective determinations are also useful for checking out the IF amplifier chain. Suppose, for example, that a particular set has weak volume but can receive all local FM stations. If the set has a tuning meter across the FM detector, note whether it is registering a normal signal level. If it is, check the stereo indicator lamp. If it is illuminated, the defect probably is in either the composite signal path or in the audio amplifier chains. If the tuning meter indication is normal, yet the stereo lamp is not illuminated, check the coupling between the detector and the input of the multiplex section.

The tuning meter often is electrically located in some circuit other than the detector. In some modern hi-fi receivers it is placed electrically in one of the early IF stages. In such cases, it can help the technician determine which end of the IF strip to check first. (However, if the meter itself is defective, the preceding test can be misleading.)

One effective method for locating a defective IF amplifier stage is to signal trace it with either a very sensitive oscilloscope or an aural signal tracer. However, when using such an instrument, be sure the demodulator probe is designed for the IF frequencies. Many probes that are suited only for AM servicing will seriously detune 10.7-MHz IF transformers.

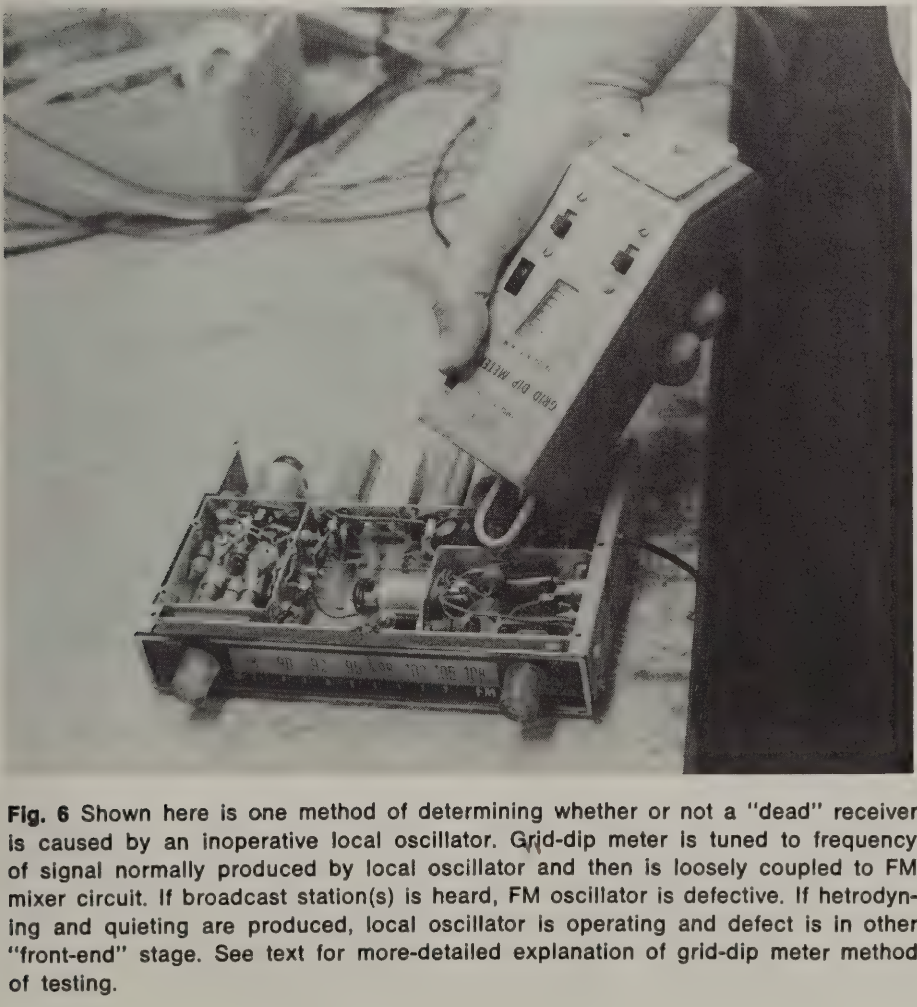


Fig. 6 Shown here is one method of determining whether or not a "dead" receiver is caused by an inoperative local oscillator. Grid-dip meter is tuned to frequency of signal normally produced by local oscillator and then is loosely coupled to FM mixer circuit. If broadcast station(s) is heard, FM oscillator is defective. If hetrodyning and quieting are produced, local oscillator is operating and defect is in other "front-end" stage. See text for more-detailed explanation of grid-dip meter method of testing.

If local off-the-air signals are too weak to permit effective signal tracing of the first two IF amplifiers (as is often the case if a low-sensitivity scope is used), try feeding the 10.7-MHz output of a sweep generator into each IF stage in succession while monitoring either the output of that particular stage or the output of the entire IF chain. Monitoring the entire chain offers the advantage of three extra stages of amplification. This added gain will be needed if "low-gain" test equipment is used.

The detector circuit in an FM set can be frustrating to troubleshoot. Certain designs will exhibit almost normal voltage and resistance readings, yet will fail to demodulate the FM signal. Analysis of detector waveforms can be useful in determining which detector component is defective.

The specific type of detector circuit being used can be another factor in troubleshooting. IC quadrature detectors, for example, are very sensitive to B+ line transients and momentary short circuits. The author recently heard an older technician complaining about this very problem. He remarked that in the older tube-type hi-fi's you could get away with being pretty sloppy with tools and what they touched. Then came transistor sets, in which sloppiness can cost a two-dollar bill. In integrated-circuit designs, such sloppiness can cost more than ten dollars. To make matters even worse, IC lead wires are spaced extremely close.

The oscilloscope seems to be the best instrument for troubleshooting just about any type of FM detector circuit. The new 10-to-12-MHz triggered sweep scopes permit you to look at the FM IF signal before it is demodulated. When using a narrow-band scope, however, you must connect it to the detector through a demodulator probe at the detector input or a low-capacitance probe at the detector output. Either way, you will be looking at the audio component. The set probably will have to be tuned slightly off station before the audio can be extracted with a demodulator probe.

This is because most probes are AM detector types, which use slope detection for FM.

The scope also is the most logical instrument to use for troubleshooting the multiplex section of the stereo FM receiver, because the majority of signals in this section are above the range of human hearing.

When confronted with a dead, or apparently dead, multiplex section, make a few preliminary observations before disassembling it. Note, for example, the amount of background hiss between stations, and whether the fringe stations usually heard are actually being received. This will indicate whether the sensitivity of the set is up to par. For example, a set with one defective IF amplifier can pass enough signal to drive the high-gain audio amplifiers, but not enough to overcome the multiplex threshold.

It is also wise to make a quickie separation check with an oscilloscope, to determine where the fault is located. This is accomplished by feeding the right channel into one input of the scope and the left channel into the other scope input. If the trace is a slanted, fuzzy line, either the set is tuned to a mono station or it is not separating the two stereo channels. A trace that resembles a nest of wiggly worms will be produced if good stereo separation is present. This information will tell you whether to start in the stereo-indicator lamp circuit or the multiplex section.

The audio chains used in FM stereo normally require only the standard troubleshooting procedures. Square-wave testing, for example, has been especially useful. Fig. 7 shows some of the ways that an audio amplifier can distort square waves.

A multiplex generator and a harmonic distortion analyzer are two other instruments that are very useful for stereo FM troubleshooting. Both of these instruments, when properly used, can save a lot of time.

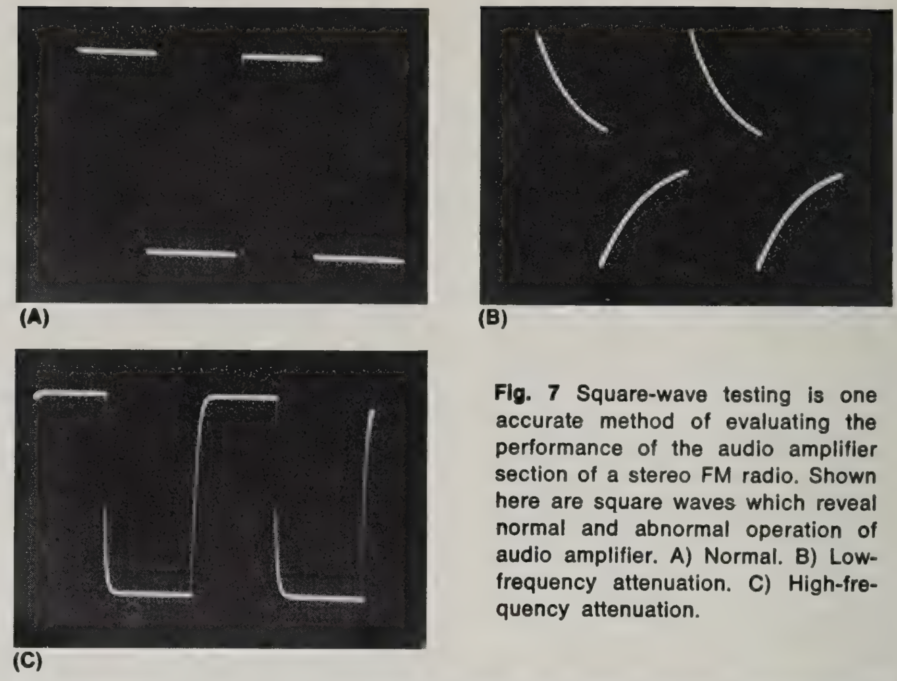


Fig. 7 Square-wave testing is one accurate method of evaluating the performance of the audio amplifier section of a stereo FM radio. Shown here are square waves which reveal normal and abnormal operation of audio amplifier. A) Normal. B) Low-frequency attenuation. C) High-frequency attenuation.

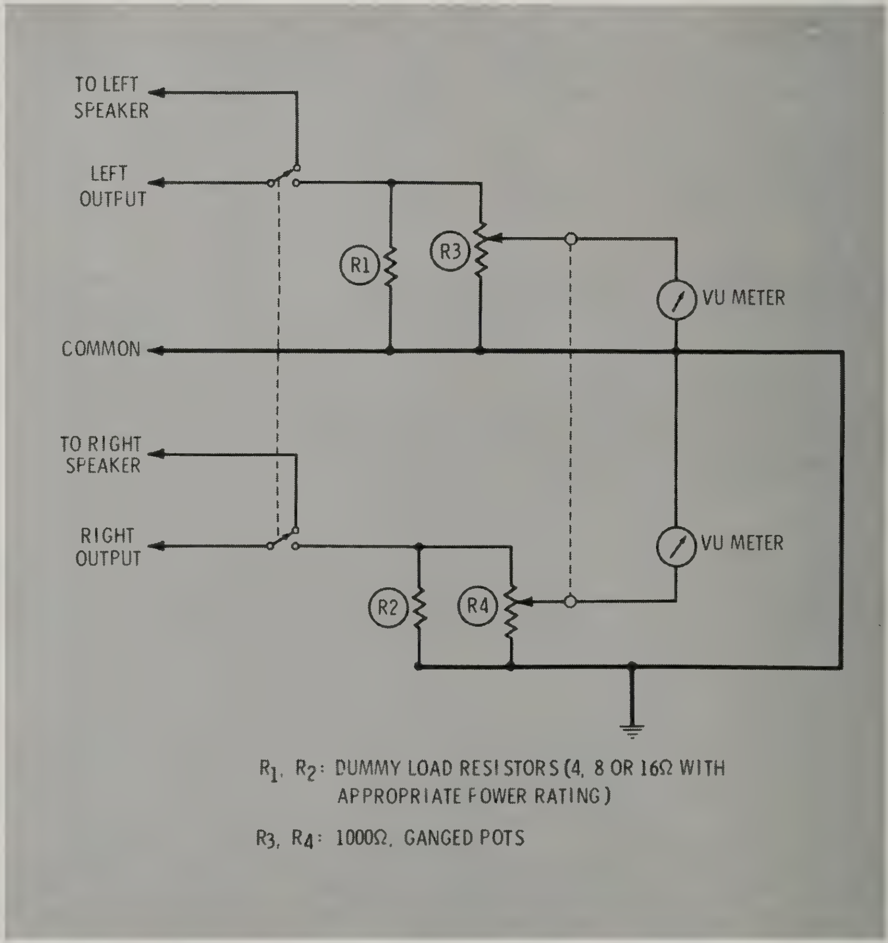

TO LEFT SPEAKER
LEFT OUTPUT
R1
R3
VU METER
COMMON
TO RIGHT SPEAKER
RIGHT OUTPUT
R2
R4
VU METER
R1, R2: DUMMY LOAD RESISTORS (4, 8 OR 16Ω WITH APPROPRIATE POWER RATING)
R3, R4: 1000Ω, GANGED POTS

Fig. 8 Test setup for checking "serious" stereo separation problems. See text for details about use.

The test setup shown in Fig. 8 is found on many hi-fi test benches. It makes possible separation checks of the entire system. Simply feed the signal to one of the stereo channels and calibrate the volume meter on that channel to 0 dB. The separation, in decibels, is read on the other meter. Although this setup doesn't take residual noise into account, it is practical for spotting serious separation problems. The Vu meters can be the "under-five-dollar" variety offered by many mailorder suppliers. ▲